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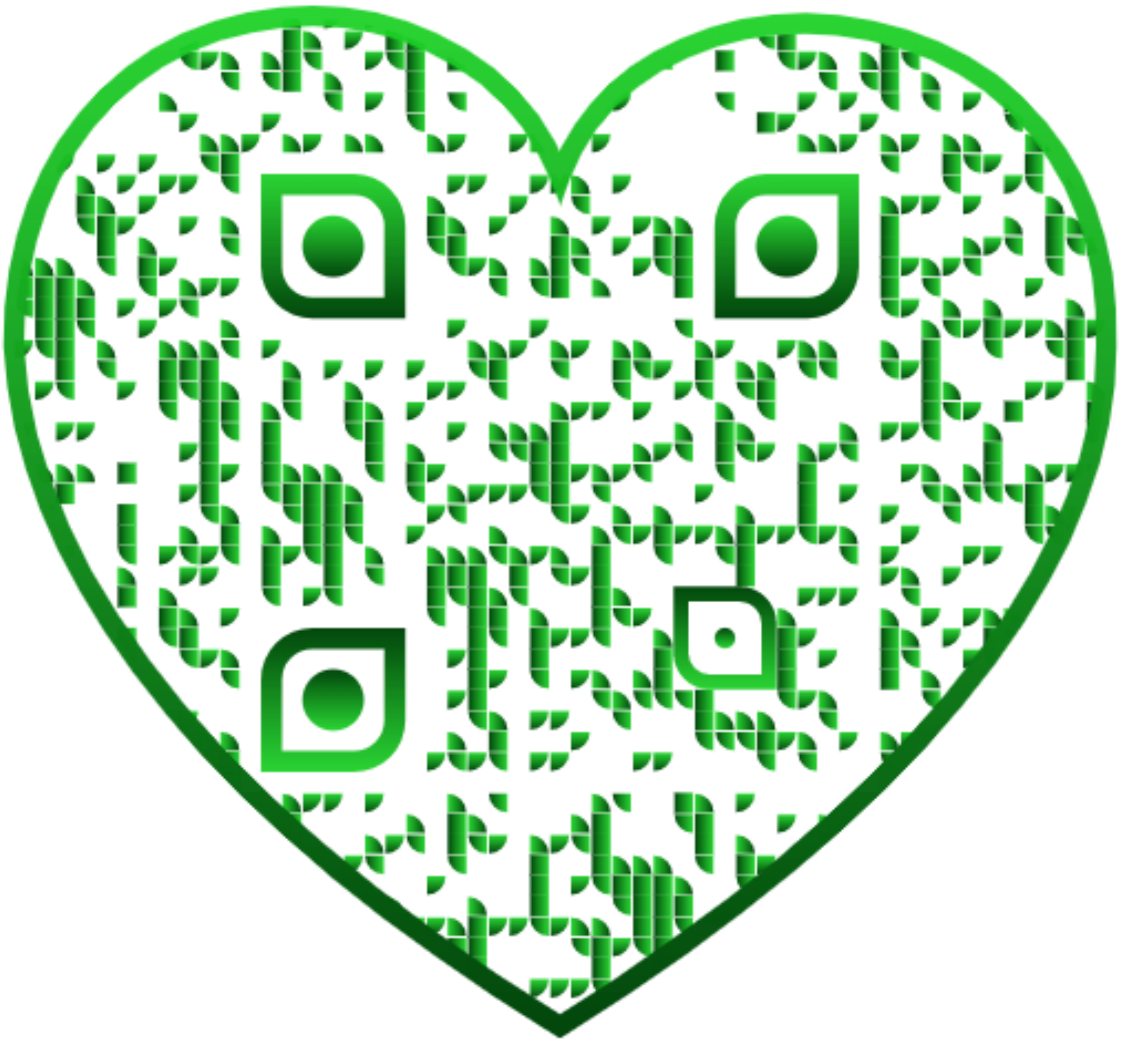

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TRUST YOUR GUT

Does knowing about production change the way we engage with music? Definitely. And on the whole, I think that's a good thing. Above all, it means we can find something to hold our attention in absolutely any song. The melody might be forgettable, the lyrics clunky, the vocal performance below-par, but if you have an interest in production, there's always something to pique it. How did they get that bass sound? Is that the original sample, or have they re-played it? Why is the snare off-centre?

At the same time, though, I think there is also a down side. The expression 'I don't know much about art, but I know what I like' is widely mocked, but it captures something that's quite important about how the majority of people experience music. Preferences are formed unconsciously, and decisions about what's good or bad are made unreflectively. People either like something, or they don't. Whether or not it makes innovative use of wavetable modulation is not taken into account.

The deeper we dive into the technical side of music, and the more analytical we become, the harder it is to recover that instinctual reaction. We lose touch with the immediate emotional response that says 'Yeah, this is great!' or 'Yuck, I don't think much of this.' And what separates the really great producers from the rest of us is that they are able to recover that naïve perspective when it's needed.

We can all tell when a vocal is out of tune or the drummer is speeding up. A much rarer skill is the ability to know when it matters. When we're working on our own music, things quickly become over-familiar, and we start second-guessing ourselves. When we work with others, the pressure is on to be positive all the time, because the producer is supposed to be supportive and encouraging.

Great producers can see the bigger picture, but more than that, they are able to influence it. They have the awareness to know when their instinctual reaction needs to be acted on, the insight to know what the appropriate action is, and the authority and the charisma to convince others to go along with them. It's easy to take responsibility when you're agreeing with someone, much harder when you need to change someone's mind, or take a decision that no-one else can see the need for.

The analytical and technical skills that we develop through experience are essential. So too are 'people skills'. But ultimately, they are all in service of something else. Call it what you like — taste, judgement, gut feeling, instinctive reaction — but ignore it at your peril.

Sam Inglis
Editor In Chief

"Great producers can see the bigger picture, but more than that, they are able to influence it."

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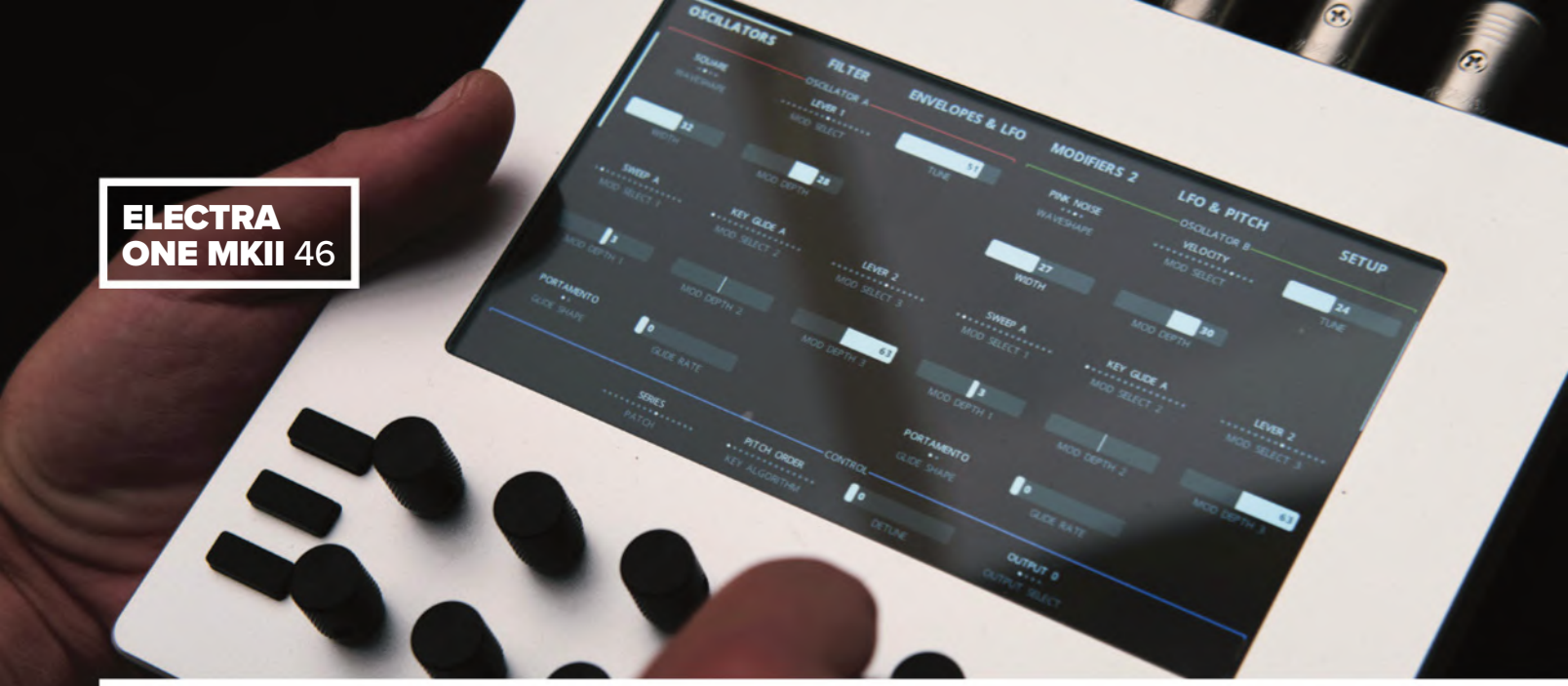

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Earthworks DM6 Capacitor Microphone

Can Earthworks' new kick mic provide a secure foundation for your drum sound?

SAM INGLIS

This may be heresy, but I don't always feel that mic choice is a make-or-break decision. What mic you use is often less important than where you put it. However, if there's one exception to this, it's the big round thing that we inexplicably call a kick drum. Despite endless variety in terms of tuning, drum size, type of heads and so on, mic choice always seems to make a big difference — and if you get it wrong, no amount of EQ ever seems to make it right.

Over the years, I've tried most of the usual suspects, and I've never quite achieved bass drum nirvana. Mics not specifically designed for the role tend to sound boxy or flabby, whilst dedicated kick-drum models either scoop the sound too much, or make every bass drum sound the same. So I was intrigued to see whether Earthworks' new DM6 could bring a smile to my face.

Lining Up

Like all Earthworks mics, the DM6 is a small-diaphragm capacitor mic, but it eschews the conventional 'pencil' shape for a short, stubby form factor that somehow looks more appropriate to its intended role. The construction is ironclad, with a heavy-duty black grille and a housing made from Earthworks' trademark polished steel. It comes with the superb Triad-Orbit M2-R ball-and-socket mount, which allows the mic to be rotated on any axis and clamped securely in place.

The DM6 is officially a supercardioid mic, and has a deliberately low sensitivity of 1mV/Pa that befits its likely use in high-SPL contexts. The published polar plot suggests a supremely well-behaved directional response at

frequencies from 125Hz up, while the frequency response chart shows a shelving rise below 100Hz that is about 9dB up at 30Hz, a broad dip in the midrange which is about 8dB down at 300Hz, and a wide, flat presence boost extending from about 2kHz to 10kHz or more. Earthworks say that the curve "represents typical frequency response, as experienced with DM6 positioned just inside bass drum port, diaphragm aligned with resonant head". This doesn't mean it was measured inside a drum, just that the measurement was taken at less than the standard 1m, reflecting the fact that this mic will typically be used close up. Self-noise is listed as 20dB (A-weighted), which is very much a non-issue inside a bass drum, and standard 48V phantom power is required. Unlike some Earthworks models, the DM6 doesn't max out the available phantom current, demanding only a reasonable 6.5mA.

Slide Into My DMs

Earthworks suggest placing the DM6 in a port hole, with the diaphragm roughly parallel with the resonant head, and that was mainly where I used it during the review period. If you want to go further in, the mic itself is compact enough, but having the mount sticking out at right angles can make threading it through a small port awkward. In practice, it's quite forgiving of exact placement, and its well-behaved supercardioid pattern means there's rarely any need to shove the DM6 right into the drum in search of better isolation. I also tried it as an 'outside kick' mic, in which role it worked nicely.

The review DM6 arrived in time for a weekend session with a band, so I threw it straight into action in the recommended spot. In the interests of science, I then swapped it out for my much-loved beyerdynamic M88 in the same position (which would be my usual choice). The difference was not subtle. Most notably, the DM6 seemed to capture an extra octave at the low end that was simply missing on the dynamic mic, yet unlike other 'sculpted' mics I've tried, it remained punchy and fast, with no overhang or ringing. It hollowed out the midrange to pretty much exactly the right extent, and made the click of the beater audible without over-emphasising it.

In short, then, the DM6 pulls off the neat trick of delivering a 'mix-ready' sound whilst simultaneously sounding natural, inasmuch as that term is appropriate for sound captured somewhere you'd never want to put your ears. Sonically, Earthworks have steered the narrow path between box and basketball, and I think the DM6 will be especially valuable for live-sound engineers. It's a bulletproof, universally applicable bass-drum mic you can put up in the knowledge that it'll sound good on any kit with minimal intervention at the desk. If this isn't kick-drum nirvana, it's certainly a new plane of enlightenment. ■■■

summary

A classy kick-drum mic that offers a mix-ready yet versatile sound with minimal fuss.

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Toontrack EZdrummer & EZkeys Expansions

Dark Industrial EZX & EKX

Toontrack get gritty with their Dark Industrial expansions for EZdrummer and EZkeys.

JOHN WALDEN

Toontrack's expansions for both EZdrummer and EZkeys offer an abundance of flavours for conventional sounds, be that for acoustic drums or acoustic and electric pianos. However, there are also some less conventional options and the most recent additions — the Dark Industrial EZX and EKX libraries — clearly fall into this category.

E To Z

Let's start with the EZX. In keeping with the steampunk-esque graphics, the expansion provides over 400 individual sounds originally sourced from acoustic and electronic drums, Foley sounds, effects and (as far as my ears can tell) a bunch of objects being hit. These have then been processed, re-amped and remixed in various combinations. The sounds are grouped into various types including kicks, snares, toms, subs, claps, atmospheres, rim shots, effects, objects, clicks and various cymbal types. Some 31 ready-to-roll mix presets — wonderfully left-of-centre and experimental — have then been built from these and you can, of course, build your own presets.

The EZdrummer UI provides some very neat additional sound design options. For example, in addition to the usual kit piece channels, the Mixer panel includes channels for jangles, clicks, effects, subs and so on, and a very effective Parallel Compression channel that works brilliantly to add some additional beef. The Mixer's effects options include Dirt, Filter, some global Pitch/Envelope options and the suitably quirky Ambience Warp knob that lets you add some additional 'glitch' to the mix. All of these can be automated to add some movement. The package is rounded out by a MIDI groove collection with grooves designed to showcase to each of the kit presets but, as the preset sounds are laid out in a similar fashion to a standard EZdrummer acoustic kit, you can easily make conventional MIDI grooves work with these sounds.

The Dark Industrial EKX brings the same sort of philosophy by blending and processing combinations of acoustic piano and organ sounds with those from electronic and synth-based sound sources. Some 80-plus hybrid presets have then been classified into four categories: keys, pads, leads/basses and textures. Again, you get some dedicated MIDI grooves with options designed to work with each of these sound categories. A little more care is required here if you want to explore using other EZkeys MIDI with these sounds (a pattern from your Acoustic Pop library might not be the most appropriate choice to trigger a Dark Industrial texture, for example).

In Use

The sounds themselves are cool and quirky, and would work in gritty electronica, industrial rock/metal or soundtrack contexts (there are some *Blade Runner 2049* and *Stranger Things* vibes to be had) and, as within the EZX, via the pop-out Effects panel in the main Keys display, you get sound design options specific to this expansion. These include envelope shaping and the ability to adjust the blend between the synth and noise elements of the sound, as well as to modify the initial character of the transient via the Impact control. Finally, the Sound Shaping section provides low-/high-pass filtering, saturation/distortion (via the Color control) and Ambience. The exact configuration varies between presets. Again, all these controls are easily automated so you can tweak them in real time to add further sonic movement to your performance.

As the Dark Industrial title suggest, the sounds within both of these expansions are intended to be more experimental in nature than some of Toontrack's more conventional offerings. However, if your next musical production is more Nine Inch Nails or Depeche Mode than Ed Sheeran or Taylor Swift, then a little Dark Industrial will be right on the money. Of the two, I suspect the EZX will perhaps have somewhat broader appeal as the hybrid drum sounds, while great on their own, also work brilliantly when layered with a standard acoustic drum kit to lend an extra dimension. However, used together, these themed expansions provide a very inspiring sound set for realising all those slightly twisted tunes in your head. Very dark and suitably industrial, if your music has an experimental streak, the Dark Industrial EZX and EKX are genuinely fun to use. ■■■

summary

Toontrack's Dark Industrial EZX and EKX provide an inspiring sonic palette with which to enjoy your inner musical darkness.

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iZotope Trash

Multiband Distortion Plug-in

A much-loved creative distortion plug-in is back from its sabbatical, and it looks like the break has done it good!

JOHN WALDEN

If you have a long-standing love for creative distortion, it's very likely that at some stage over the last 10-plus years you'll have tried iZotope's Trash 2. That fan-favourite was discontinued quite some time ago, but iZotope have now resurrected the Trash concept. The new version is not Trash 3, though. It's called, simply, Trash. But it's very much a new product, with a much more contemporary feel than previous versions. As well as the full desktop version of the Mk3, there's a free Trash Lite version and, for those who make music on an iPad, there are separate AUv3 Trash and Trash Lite plug-ins available in the App Store. So, whether you are a veteran Trash aficionado, or just looking to add some further creative sound-mangling options to your sound design, is the new Trash worth a bash?

Brand New Trash

As can be seen in the main screenshot, the Trash has a modern GUI that's consistent with other products in iZotope's catalogue, and with all the controls contained in a single-page display, it promises to be quick and easy to use. To some extent, that's been made possible by some careful focusing of the feature set. Two main modules, Trash and Convolve, now dominate the central part of the display, and as well as the ability to turn these on/off individually you can also switch their order in the signal chain.

At the top are the usual preset access (including a random preset selector), bypass and global wet/dry mix controls. There's also a spectrum display, through which the user can configure Trash to operate over one, two or three frequency bands, and it's worth noting that if you opt for a multiband setup, each band can have independent Trash and Convolve configurations, and that means you can process your lows, mids and highs in entirely different ways should you wish to. In the bottom third of the display, you can access compact control sets for the Envelope, Filter and In/Out panels, and the last of those include useful auto-gain and limiter options to avoid signal overload.

At this point, some users of Trash 2 (those who went beyond just loading and tweaking presets) may well be wondering whether this modern take on Trash is, if anything, a little *too* streamlined. For example, the multiple EQ stages, multiband compression, waveshaper, delay and multi-module signal-chain editing options of Trash 2 are not included here. It's definitely a very different design philosophy – a more focused feature set and streamlined workflow – which is probably why iZotope are not positioning this new release as Trash 3 but rather as 'Trash reimagined'. It might leave some hardcore Trash 2 users a little disappointed, but I suspect that the new Trash will appeal to a new and wider audience.

The Style panel provides ways to customise many of the distortion algorithms in the Trash module.

Appetite For Destruction

In the new Distortion module there are over 60 different distortion types to choose from, and the user can load up to four different distortion algorithms simultaneously, one into each 'corner' of the X/Y pad. The distortion types are organised into multiple categories (Distort, Drive, Faulty, Fuzz, Heavy, Retro and Saturate) so the vast choice never threatens to overwhelm, and they range from the subtle to the outright insane. Using the red node on the X/Y display, you can adjust the blend between the four distortion styles, either as a static setting or dynamically, using automation or a control surface. The large Drive knob controls the input gain (and, therefore, how hard you push the chosen distortion algorithms), while the Mix knob controls the blend of dry/processed signal in the Trash module.

A Tilt EQ (broadly speaking a high shelf cut/boost coupled with a low shelf in the opposite direction) lets you shape the sound before it hits the distortion processors, and you can set the tilt's centre »

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» frequency as well as the amount. Further options tucked into the GUI include a dice icon at the top of each X/Y pad: a means of randomly selecting (in this module) the distortion algorithm selections. In addition, a small 'slider' icon (bottom centre of the X/Y pad) pops open Style controls for the four distortion algorithms, allowing you to adjust the intensity of the effect.

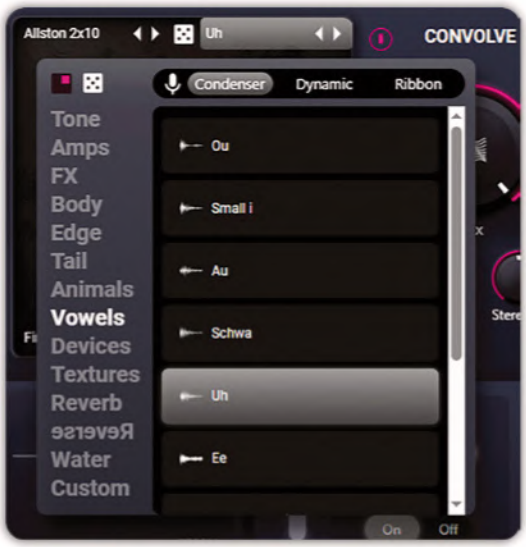
You can manual automate the X-Y pad node position, but there are also some modulation sources in the Trash module. First, the Envelope panel settings can be used to modulate the Drive knob. The Envelope panel is simple and intuitive, with modulation amount, attack and release controls, but a key to note is that it is based on an envelope follower, so the stronger the input signal, the greater the potential modulation. Also, the Drive knob's modulation amount control offers both positive and negative modulation values. Second, you can engage the modulation node (using the button located to the top and right of the X/Y pad). This cyan node can then be positioned on the pad to create a modulation path to the red node, and movement along this path is also triggered by the envelope.

The envelope can also be used to modulate the filter frequency, of course. The filter itself is a fairly simple resonant low-pass style, but with Scream letting you control the filter resonance (with self-oscillation possible at higher settings) and Heat adjusting how hard the filter is driven, you can use it to inject plenty of sonic colour into a source.

Acting On Impulse

The Impulse Response module provides a similar X/Y pad experience but this time you can load up to four different impulse responses, again one placed in each corner and with a node that lets you morph between them all. There are over 600 impulse responses provided and these are organised into various categories, including Amps, FX, Textures, Tones, Vowels, Reverbs, Reverse and (yes, really!) Animals, with multiple microphone options for some of the IRs. In addition, each instance of Trash lets you load a single audio file up to one second long to use as your own 'custom' impulse response (drum hits can be fun...). The Mix, Width and Stereoize controls then let you adjust the dry/processed balance and stereo image created by the overall processing.

As with Trash 2, the Impulse module makes Trash rather more than 'just a distortion processor' and, whether used with or without the Trash module, there's plenty to explore. You can place your sound into a real or (using some of the more unusual IRs in the Reverse, Devices and Water categories) otherworldly type of space.



While distortion might be Trash's headline, don't underestimate the sound design potential offered by the multitude of options within the Convolve module.

Trash Talk

This reimagined version of Trash is undoubtedly much more intuitive to use than its predecessor was and, whether the Trash 2 faithful agree with the direction of Trash travel or not, I suspect this new version will attract a lot of new users who would have found the original somewhat intimidating. In any case, as demonstrated so ably by the presets, this new release is very capable of serving up anything from super-subtle saturation to insane and over-the-top sonic destruction. And, by the time you have all three frequency bands in operation, each with their own distortion, IR and modulation settings, things can still get pretty deep — it still has that amazing ability to transform even the blandest of sources into something totally new that could inspire your next track. Indeed, you can consider the box for Trash's primary purpose — a creative tool for distorting and transforming your sounds — well and truly ticked. Whether it's drums, bass, synths, vocals, or some other sonic element of your project that needs a touch of grit, or total destruction, the new Trash still delivers.

From a practical perspective, there are perhaps only two items on my own 'would have been nice' wishlist. First, you're limited to one custom IR, and being able to load them in every 'corner' would have been nice. Second, I think a band-level preset system would have been handy. But these aside, I think iZotope's approach to the redesign of Trash is both astute and pragmatic. Packing so much power into such a simple GUI is no mean feat.

Trash is sensibly priced already, but existing iZotope customers also get the benefit of upgrade and loyalty discounts, and there's also the free version too. While some may not wish to put out the old Trash, I suspect lots will be happy to bring in the new Trash. Hats off to iZotope: Trash 2024 is lots of fun and very accessible in use. ■■■

With over 60 distortion algorithms distributed over seven categories, Trash lets you go from subtle to insane with relative ease.

summary

iZotope's radical reimagining of the Trash concept has delivered us a very capable yet easy-to-use creative distortion plug-in with a stylish GUI and plenty of sound design potential.

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- Dual-Force™ Technology
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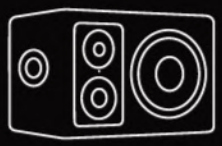

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Universal Audio SC-1

Modelling Microphone

UA's new range brings mic modelling to the masses!

SAM INGLIS

When Universal Audio acquired mic modelling maestros Townsend Labs, the obvious intention was to deploy their Sphere technology in more affordable, simpler products. The first step along this road was last year's Sphere LX, and UA have now gone much further.

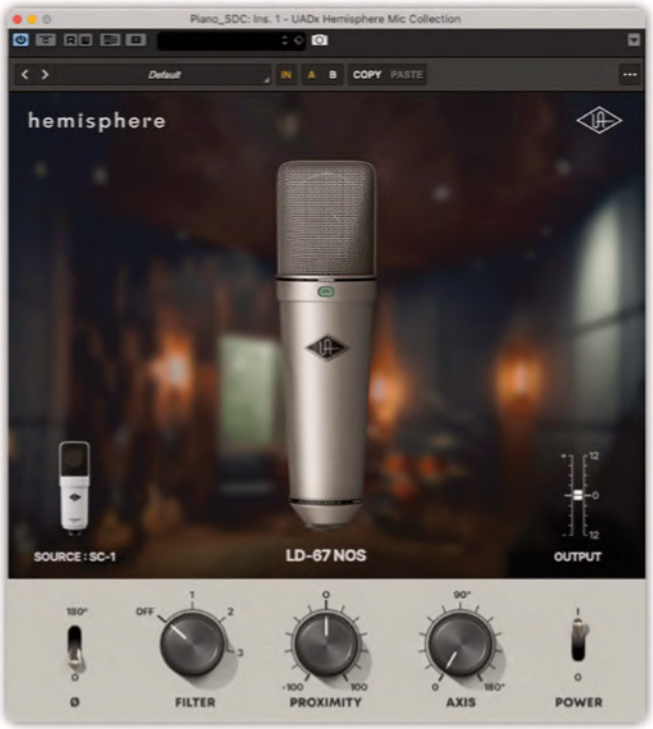
The Sphere DLX and LX have separate outputs for the front and rear sides of the capsule, allowing the polar pattern to be manipulated in the Sphere plug-in, but requiring two matched preamp channels. The new generation of UA modelling mics are fixed-cardioid designs that connect to a single preamp input, just like any other mic. There are separate large-diaphragm, small-diaphragm and moving-coil models, plus a cut-down version of the Sphere plug-in named Hemisphere, which can run natively or on UAD hardware DSP platforms.

Depending on which source mic you have, Hemisphere delivers a curated selection of modelled mics that use the same operating principle. For the SP-1 pencil mic, that means a Neumann-heavy catalogue including the KM54, 56, 84 and 86, plus an AKG C451/CK1, whilst SD-1 dynamic mic owners get a virtual Shure SM7, Electro-Voice RE20, RCA 77DX, beyerdynamic M88 and Sennheiser MD441. The large-diaphragm SC-1 was supplied for review, and offers the widest choice of all, with Neumann U87, U47, U67 and TLM103 models joined by a Sony C800G, Telefunken ELA M 251 and AKG C12 and C414 B-ULS.

Physically, the SC-1 is a smart, distinctive and reasonably compact mic. It boasts a self-noise of 12dBA and, notably, the ability to tolerate SPLs up to 145dB. It comes with a standmount rather than a shockmount, but will fit a Rycote USM or similar if necessary.

Rounded Out

Some Sphere features depend on that mic's dual-output capability, and others probably go unused in most recording contexts. Hemisphere thus does away with controls such as off-axis correction and compensation for the effects of portable recording booths. It also lacks the Sphere plug-in's ability to mix and match two virtual mics within the same window, though there's nothing to stop you duplicating a track and using a second instance of Hemisphere. So, along with output level trim and polarity controls, you just get a three-position filter switch, and dials for Proximity and Axis. And, in all honesty, that's all you need for most situations.



I wasn't able to compare the SC-1 directly to a Sphere, but I did put it up against my U87 and C414B-ULS, and in both cases it got very close to the originals on close-miked vocals, the main application for this product. In fact, although I could probably tell the SC-1's U87 model apart from the real thing in a blind test, I certainly couldn't tell you which of the two I preferred. It sounded noticeably more like my own U87 than Warm Audio's WA-87 MkII did in the same test. All the other models sounded like I'd expect the real thing to sound, too, with the C12 and 251 offering different flavours of airy brightness and the 67 and 47 producing complex, rich, mid-focused sounds. I'd rather have a C414 EB or Sony C37 than a virtual TLM103, but between the eight options on offer, you should be able to find at least one that works on any vocal.

The Sphere is a remarkable product, but its immense versatility comes with a learning curve and a price attached, and is unlikely to be needed in a straightforward vocal tracking session. The SC-1 and Hemisphere deliver exactly what most people need most of the time, in a good-sounding package that is child's play to use. It's flexible enough for probably 90 percent of real-world applications, and affordable enough to open up a whole new market for UA's modelling technology.

summary

Effectively eight mics in one, UA's SC-1 is an affordable and friendly introduction to the world of mic modelling.

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Electro-Voice Everse 12 Active PA Speaker

E-V pack their battery-powered PA tech into a 'full-size' speaker.

PAUL WHITE

Combining Electro-Voice's loudspeaker technology with Dynacord's design expertise in amplifiers and DSP, the two-way Everse 12 is the big brother to the Everse 8 we reviewed in January 2023, employing a similar setup but with a 12-inch woofer running alongside its wave-guide-mounted, one-inch, ferrofluid-cooled titanium-dome tweeter. Once again, the superpower of this speaker is its ability to run from its own rechargeable lithium-ion battery, giving up to six hours of use at high volumes or around twice that with more moderate use. When no signal is present, a power-saving mode preserves the battery charge, and the battery charges whenever the speaker is plugged into the mains, even when powered off.

Moulded from polypropylene and also available in white as an alternative to the more usual black, the ported cabinet is shaped so as to allow it to be used vertically as a 'kick-back' floor speaker, in addition to the more usual sideways monitor orientation or indeed as a 'main' speaker, mounted on the pole socket fitted into the base. A comfortable carry handle is moulded into the top of the cabinet and a powder-coated steel grille meeting IP43 requirements for resistance to light rain and splashes protects the drivers. This speaker is well suited to portable use as well as pub gigs, with a weight of 14.16kg and a compact size of 345 x 597 x 347mm. As shipped, a waterproof plastic cover protects the rear panel. This is quite a tight fit, so may need to be levered off using a screwdriver to save your fingernails.

> "The sound quality on offer here is impressive."

Bluetooth music streaming is built in, and Dynacord's QuickSmart DSP handles driver protection, effects, EQ, anti-feedback and crossover duties. Remote control is also possible over Bluetooth using Electro-Voice's QuickSmart mobile app for iOS and Android. Everse speakers can also be used in pairs for stereo playback over Bluetooth.

The effects section available to channels 1 and 2 includes compression and ducking in addition to the usual delays, reverbs and so on (these are arranged as 30 presets). A limiter helps keep peak levels in check, and when this is triggered, a Limit warning appears on the display. A nice touch is the provision of a 12V DC port that can power an Electro-Voice RE3 or R300 wireless receiver — or indeed any third-party receiver that runs on 12 Volts. There's also a USB-C port that supports device charging. On mains power the speaker runs from a standard IEC connector.

By way of specifications, the speaker can produce a maximum SPL of 126dB at one metre, covering the frequency range 45Hz to 20kHz (-10dB) or 55Hz to 20kHz (-3dB). Total power handling is quoted as 400 Watts peak. Dispersion is 100 degrees horizontal by 60 degrees vertical, the asymmetry helping to make best use of the available power. Dedicated master settings optimise the EQ to the orientation of the speaker. There's also a choice of high-pass filter frequencies for use with an optional sub.

Panel Games

Functions are selected and edited using a single turn-and-push encoder that also functions as the master volume control. An illuminated button over each input selects it for editing, with the LCD showing parameter names, parameter values and battery status. With no buttons lit, the master settings are active. The monochrome display is functional and clear, though as it's on the back of the speaker, you may have to bend down to see it unless the speaker is mounted on a tripod. Pairing with the app is also straightforward.

The mixer section features two combi XLR/jack inputs that can accommodate either mic or line-level signals, with switchable 48V phantom power on input 1 and high-impedance instrument compatibility on input 2. Ducking, if used, is implemented in such a way that either channel 1 or 2 may be designated to control the levels of the remaining inputs. For channels 3+4, a 3.5mm aux »

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» input jack accepts audio from other devices. Alternatively, if you want to stream audio over Bluetooth, this will appear on channels 3+4. If only a single Everse speaker is being used, then anything fed into channels 3+4 will be made mono. There's a jack for connecting an optional footswitch, which can be used to turn the effects on or off. A balanced mix out on XLR facilitates connecting to other powered speakers.

The AFS feedback suppressor uses 12 extremely narrow notch filters that lock onto and cut the feedback frequencies when feedback is detected, allowing for a little more gain without annoying ringing. These are best set up during the soundcheck by 'ringing out', though leaving some filters unallocated can help avoid problems during performance, especially if the vocalist is in the habit of moving around while holding the mic.

There's no shortage of onboard effect presets, though the reverbs and reverb-plus-delay combinations are probably the most useful for vocalists. Different levels of the same effect can be added to channels 1 and 2. The effects section also includes modulation effects and doubling, and for those looking for an easy life, the mixer's bass/mid/treble EQ (+6/-12 dB gain range) comes with presets for voice, or instruments such as guitar. A separate eight-band parametric EQ is available on the main output allowing for overall room correction, and there's a choice of presets that tailor the sound for Music, Live, Speech or Club environments. It is also possible to dial in a delay for use in time-aligned systems where the speakers are a long way apart.

E-V's QuickSmart app provides remote control over the Everse 12's DSP and mixer.

All of the Everse 12's functions can be operated from the rear panel. The cabinet itself is angled such that it can be used as a stage wedge, a 'kick-back' speaker, or a traditional PA.

Remote Control

The QuickSmart app is available for iOS and Android devices, and in Mixer Edit view, features tabs for X-Over, Location, Function and Mode. Selecting Edit Mixer gives a choice of Main or effect settings. The effects section can be bypassed from the Effects page. If the mixer is set to Basic view, the master volume and EQ options can be accessed, along with speaker orientation presets and sub settings. I found the app clearly set out and easy to use.

Sound Check

As with the smaller version we reviewed last year, the sound quality on offer here is impressive. So many box speakers are all about the highs and lows but disappoint in the important midrange, but there's good definition here. There's more heft to the sound and a little more overall level compared with the Everse 8s, but at the expense of around twice the weight. Even so, the speaker is still an easy one-handed lift. It would probably be wise to add a sub if you intend to put instruments such as kick drum or bass guitar through the system at any kind of volume, but for general PA use, the Everse 12 speakers sound really clean. Solo artists or duos playing smaller pub gigs could easily get by with a single Everse speaker.

The battery powering option might get you out of a bind in a power cut, but really comes into its own for outdoor events such as garden parties, village fetes and serious busking. In my tests I couldn't detect any difference in performance between running from mains or from the battery.

The quality of the onboard effects is impressive; there are a few that are less likely to be used in a typical live scenario, such as the very long reverbs, but there are enough useful options that work well for vocals and acoustic instruments. The anti-feedback system also proved useful in eliminating offensive ringing. If you are using an external mixer, you can turn off all the onboard effects and processing in the master settings menu.

While there are cheaper two-way 12 inch 'box' speakers, this one comes with the Electro-Voice pedigree and is clearly very well engineered. It sounds clean, it plays reasonably loudly and it offers a wide enough selection of effects to meet most needs. While battery power adds to the price, it does mean the you can enjoy uncompromising performance away from a mains power supply when you need it.

summary

The Everse 12 is portable, delivers high-quality sound with a good range of effects and has a battery powering option for when you need it. The control app is also easy to navigate.

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UNIVERSAL AUDIO

Endorphin.es Ghost Pedal

Multi-effects Pedal

Forged in the world of modular synthesis, Endoprhin.es have taken their first, bold stride into the effects pedal market...

WILLIAM STOKES

Barcelona-based Ukrainian developer Endorphin.es first collaborated with Andrew Huang last year on the Ghost Eurorack module, a creative audio processor that we're told is "based on [*Huang's*] modern music production techniques" and boasts spatial effects, dynamics processing, filtering and more. Almost a miniature 'system within a system', the module allows for flexible routing between its many processors at the push of a button, and on top of that they managed to cram everything onto a 16HP faceplate.

With the module proving a success, Endorphin.es and Huang decided to create a pedal-format Ghost, and that's a bigger deal than it might sound: this is the first time they've reached beyond the bounds of modular synthesis and into the wider world of effects pedals and outboard. With a reimagined interface, a sturdy chassis and, of course, endowed with footswitches, it's Endorphin.es as you've never experienced them before, without a patch cable in sight. Well, almost...

The Ghost Pedal's different effects and processor sections can be routed to each other in any order.

Overview

At the core of the Ghost is a 96kHz, 32-bit digital effects processor with three distinct sections: spatial effects (consisting of delay and reverb), filtering and distortion. These can then be placed, using a central button, in any order and this paves the way for a vast range of different responses. You can distort a filtered reverb, for example, or cram a reverberant distortion through a resonant filter.

The spatial effects have controls for a conventional delay (amount, time and feedback) and there are several nicely designed reverb algorithms: hall (with shimmer), reverse and spring. The filter offers modes for bipolar low- and high-pass, band-pass or harmonic-rich comb filtering. Distortion can be dialled in with Amount, shaped with Tone and then fed through a virtual cabinet. Some of the above functions are accessed using a well-thought out Shift button that, for example, adjusts resonance on the filter, tone on the delay, and even unlocks a powerful bit-crusher behind the Volume knob.

As an old-school fuzz and Big Muff fan, I'm used to accepting a hefty noise floor along with whatever grit I'm going for, but the Ghost's 8x oversampled digital distortion not only sounds very nice, it also comes without much noise to speak of, which is great. That's not to say there wasn't a little noise, nor some much odder noises from time to time (whose origin I genuinely wasn't sure of at some points!), but overall it's fair to say the Ghost is very clean. This is one of the many reasons it stands up so well in pretty much any live or studio setting.

The cabinet simulator isn't a total showstopper, but it sweeps from a small speaker on one side (summing to mono if the input is stereo) to a wide and deep dual-amp image on the other. It's a handy tool for sculpting tone and image in conjunction with the distortion, particularly when used at the end of the signal path, and to my ears it very much gets the job done, particularly when it comes to non-guitar signals — I'm a firm proponent of amping synths for extra character, so was delighted to see the Ghost working so excellently with my studio's monosynths.

Modulation & Expression

Andrew Huang (see box) states that the Ghost has "what I think is the best modulation and expression pedal assignment system ever to exist in a pedal". I'm not nearly about to throw 'ever-to-exists' around, and maybe it's Huang's charming persona, but I find myself trusting that he genuinely believes this: the attention that has clearly gone into this aspect of the Ghost is quite astonishing and, best of all, it really works. In fact, this is one of the areas in which the Ghost really excels.

A simple but effective onboard matrix has a dedicated button to route modulation to any one or more (even all, if desired) parameters. A Shape knob presents a choice between a variety of waveforms, from sine to square to stepped random and more, as well as an envelope follower. An adjacent Speed knob then determines the LFO (or LFV in the case of the random waveforms, strictly speaking) rate or the envelope follower slew.

»

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The expression pedal input doubles up as a CV input — so you can use this pedal in much the same way you might a typical synth module.

» What this all means is that the Ghost's functionality can be expanded suddenly into a litany of other pedal territories with the simplest of gestures. Route a sine wave LFO to the volume, for example, to render the Ghost a tremolo pedal. Route the envelope follower to the filter to turn it into an auto-wah. Route a smooth random wave to gently modulate a short delay time for lo-fi, tapey nostalgia straight out of the ZVEX Instant Lo-Fi Junky playbook... the list goes on. One wonders if such economic but wide-ranging applicability could only be the fruit of a Eurorack developer's lateral thinking. The Ghost's digital brain is perceivable at times — there's a slight latency on filter sweeps, for example — but, overall, its digital side is used to excellent effect, not least with different coloured LEDs providing quick and intuitive visual feedback about what's going on in various modes.

The expression pedal input opens up further options, and you probably saw these coming! Since it can be mapped to anything, once again it can turn the Ghost into anything from a volume pedal to a wah-wah, but one that's much more tunable in range and resonance than a classic wah. It can be used to speed up and slow down the LFO for expressive modulation, and so on.

Earlier I said 'almost' with regard to there being no patch cables to speak of. This is because the Ghost also does a rather excellent job of entertaining CV from your modular system via this expression input. With the onboard LFO working at one rate and an external LFO at another, at times it was almost as if the Ghost had gone full circle to behave more like a Eurorack module than a pedal. When used with the aforementioned monosynths (in my case a PWM Malevolent, replete with numerous Eurorack-compatible patch points), it was a pleasure to both feed the Ghost audio and modulation from the same instrument, as well as a MIDI Clock to sync its delay time.

On that note, in case it wasn't already clear, it's worth mentioning that the Ghost is a highly connectable beast. The rear panel offers stereo inputs (switchable between line and high-impedance inputs) and outputs, 5-pin MIDI DIN in, out/thru, as well as an expression pedal input. MIDI in can be used to send CC messages to different parameters, and also MIDI Clock, while the out/thru can send Clock as well as CC messages according to the movement of the Ghost's knobs, which is a nice touch.

There's no USB to speak of, but I didn't miss it at all: the Ghost more than stands up as a piece of studio outboard ready to be plumbed into your setup, as well as a guitar pedal, but while it presents so many options, it just about stays on the right side of simplicity to be as usable and effective as possible.

I will say, though, that the expression pedal assignment process felt a touch convoluted at times. It requires Shift+Modulation button combinations in conjunction with various pedal movements, but while this took me a couple of goes to get right at first, it's difficult to imagine things being any other way without sacrificing some of the pedal's flexibility. Further, the Ghost's nine preset slots mean that even if it takes some time to fine-tune each sound in the first instance, by the time you're in a live situation it's a breeze to cycle through them and achieve, for all intents and purposes, exactly the sound that you used in rehearsal. It's also possible with each preset to assign the pedal's Tap footswitch to one of three functions: it can be a tap tempo for the delay time, it can control the delay's looper function, it can cycle through reverb algorithms or it can choose different effect routings.

Spirited Performance?

All things considered, the Ghost Pedal is a slam dunk for Endorphin.es. They've given us very little to complain about with this multi-effects powerhouse, and it's a worthy competitor for even the vaunted pedals from Hologram Electronics and Empress. The Ghost may just herald a bold new era for its developer, and with it a newly devoted guitar-playing user base.

Andrew Huang

Andrew Huang has created two sets of explorative presets for the Ghost, named 'Heavy' and 'Experimental'. For those unfamiliar with him, Andrew Huang is a Canadian producer and 'synthfluencer', best known for his hugely popular YouTube channel, on which he hosts gear overviews, technique tips, educational insights and an array of entertaining musical experiments. It has, at the time of writing, almost 2.5 million subscribers. Each video careens through its content with the frenetic pace typical of YouTubers (that is, multiple cuts per sentence, blink-and-you'll-miss-it visual jokes and titles veering dangerously close to the 'you won't believe number four!' school of Internet clickbait). I'm writing that with more than a little tongue in cheek, though: Huang does what he does exceptionally well. Aside from being an excellent educator, he's a formidable synthesist by any standard and has upwards of 40(!) albums to his name. He professes a deep love for Make Noise instruments, having previously dubbed their Black and Gold Shared System Plus 'the world's most beautiful synth', and has numerous videos online elegantly explaining a variety of potentially daunting concepts with signature boyish enthusiasm.

In many ways, then, Huang is the ideal collaborator for a product like this. He has tested hundreds of units, is an active artist in his own right, and is so influential in the online music technology community that he can easily collate the feedback of a tranche of other well-versed reviewers and YouTubers behind the scenes. And, when the time is right, he can of course simply use his own outlets to put some serious heft behind its publicity.

summary

Better known in the modular synth community, this is Endorphin.es' first foray into the pedal market, but the quality, inventiveness and playfulness of this device suggest it won't be their last!

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With vintage Neumann KM84s becoming ever more unaffordable, does this kit-based tribute fill the gap?

SAM INGLIS

Microphone Parts SDC-84

Small-diaphragm Capacitor Microphone

Microphone Parts is one of those names that leaves you in no doubt as to what the company make! Based in California, they specialise in components for upgrading affordable capacitor mics, but also offer complete self-assembly kits and fully assembled microphones. Most of their own designs are based to some extent on classic models from the likes of Neumann and AKG, and none more so than the SDC-84. A small-diaphragm capacitor mic loosely modelled after the Neumann KM84, it's available as a kit, but you can also opt to buy it pre-built individually or as a stereo pair, and it was the last of these options that was sent for review.

What's In A Number?

Neumann discontinued the KM84 in favour of the KM184 sometime in the '90s, and although its replacement has better specifications, many people still prefer the sound of the older mic, with its transformer-balanced output stage. Unlike the 184, the KM84 was a modular microphone; you could pair a KM body with the 83, 84 or 85 capsule assemblies, which were respectively omnidirectional, cardioid and 'speech cardioid' with an inherent bass roll-off. Microphone Parts' offering can also be ordered with three different capsule types, but in this case the options available are omni, cardioid and hypercardioid. They say they have gone to considerable lengths to replicate the capsule used in the KM84, while a 'new old stock' JFET occupies the key role in the impedance conversion circuitry. There is an output transformer, too, but interestingly, it's not modelled after the BV107 that was used in the KM84 and other Neumann pencil mics; rather, it employs the turns ratio and lamination material of the BV8 transformer from the Neumann U47.

It's not intended as a slavish recreation of the KM84, then, but the SDC-84 shares some cosmetic similarities with its inspiration. The matte silver coating is a pretty good match to Neumann's, and it even sports a cheeky diamond-shaped badge — which is actually slightly annoying, since it invariably traps the elastic when you slide the mic in and out of its supplied shockmount. The SDC-84 is roughly the same diameter as the KM84, but about 50 percent longer, and the pattern of vents in the side of the body and capsule is more reminiscent of Schoeps than Neumann. The capsule contacts the impedance converter using a metal rod rather than the KM84's pin, and unsurprisingly, is not interchangeable with an actual Neumann capsule unit.

The KM84 had a 10dB pad, engaged using a rather fiddly slide switch that often misbehaves, in my experience. The SDC-84 has no switches, but the kit includes an optional feedback capacitor that pads down the output from the capsule so that it doesn't overload the rest of the circuitry when high sound pressure levels are encountered. They recommend including this capacitor in your build if you're likely to use the SDC-84s as drum overheads, or on other loud sources, but to leave it out if you want them mostly to mic for acoustic guitars and the like. Without the capacitor, the SDC-84 will typically be 15dB or so more sensitive than the KM84. The review mics had the pad fitted, and were comparable to my own KM84s in sensitivity. Some people, myself

included, like to use KM84s to close-mic snare drums; Microphone Parts don't recommend the stock SDC-84 for this, and instead offer a separate kit called the High SPL Edition.

The review pair was supplied with the cardioid capsules, but can be ordered with any configuration you choose. Mics, shockmounts and spare elastics are housed in a lightweight flightcase. Build quality was beyond reproach, and unlike many mics in this price bracket, there's nothing at all cheap-feeling about the metalwork or finish.

The Real Thing

The obvious question to ask of a mic like the SDC-84 is "Does it sound like a KM84?" But if someone asked that question, I'd be tempted to respond "Which KM84?" Like many vintage mics, the Neumann classics can be somewhat variable, especially if they've seen a lot of use, and my two sound noticeably different from one another. On the grounds that one of them has lived a pampered life and is cosmetically in mint condition, I tend to use that one for comparisons, but who knows how representative it is of KM84s generally.

If we rephrase the question as "Does the SDC-84 sound like this particular KM84?", though, the answer is a pretty enthusiastic "Yes." One of the things that most people like about the KM84 is its relatively flat frequency response; there's a hint of sparkle in the 6kHz region, but it's balanced by a dose of midrange thickness from the transformer, and there's none of the tizzy or fatiguing high end that other pencil mics can deliver. Another thing that makes it a *bona fide* classic is its smooth, well-behaved off-axis response. On both counts, the SDC-84 aped my test KM84 very closely indeed. On many sources, in fact, I don't think I would be able to tell them apart in a blind test. Both mics bring the same sense of solidity and authority to the recorded sound; it's neutral, in that it doesn't exaggerate any particular region of the frequency spectrum, but it's far from dull. The only instrument where could hear a clear difference was on my resonator guitar, where the Neumann mic seemed to bring out the complexity and richness of the midrange more effectively.

Compared with other vintage models such as the U47 and C12, there are surprisingly few modern mics inspired by the KM84. Although it's not intended as a direct clone, the SDC-84 gets as close to the sound as anything else I've tried, and even if you're not handy with a soldering iron, you can get a matched stereo pair for much less than the cost of a single original. The kit version is more affordable still, and if you're confident in your ability to populate a fairly small PCB, will give you a seriously good pencil mic for very little outlay.

summary

An attractive and affordable homage to a classic mic, the SDC-84 is sonically very close to the real thing.

$ Single SDC-84 $379 (kit) or $499 (pre-made); stereo matched pair $829 (kit) or $999 (pre-made).
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Antares Auto-Tune Pro 11

JOHN WALDEN

Pitch-processing Software

Antares look to bring a little more harmony to their iconic pitch-correction tool.

If any music technology product has earned the title 'household name', it has to be Auto-Tune. The original pitch-correction processor, it first arrived in 1997 for Pro Tools systems before making the move into hardware and then into the native plug-in world. Over the years, it has faced serious competition from other plug-ins and, increasingly, tools built into DAWs, but Antares haven't rested on their laurels. Just 15 months after we reviewed Auto-Tune Pro X (*SOS* March 2023: https://sosm.ag/antares-auto-tune-pro-x) they've delivered version 11.

It Goes To 11

Auto-Tune Pro 11 runs in Windows and macOS hosts (there's native Apple Silicon support) and supports the AAX Native, VST3 and AU plug-in formats. There's also ARA2 compatibility, so in suitable hosts there's the prospect of deeper integration with your DAW. It can be bought outright, with a single-payment perpetual licence, or through the Auto-Tune Unlimited subscription (subscribers get automatic access to version 11, along with a suite of other plug-ins). Naturally, there are upgrade options for owners of previous versions.

Before diving into this review, I'll start with the briefest of refreshers, so if you're already familiar with Auto-Tune, you can skip this section. Auto-Tune's primary purpose is described by its name, and used in its Auto mode, it delivers what is arguably the best automatic pitch-correction experience you can currently get in software. It does this with an easy-to-use control set that's surprisingly flexible. Whether you just want to give your vocal a gentle nudge in a transparent way, go all Cher-meets-T-Pain creative, or anywhere in between, you'll find a suitable preset to get you started. You can tune chromatically or to a scale. When required, Auto mode can be used in real time, courtesy of a low-latency option. And, rather neatly, if you have multiple instances of Auto-Tune active in your project, a drop-down menu at the top left allows you to switch between them — much more efficient than having to close and open multiple plug-in windows! »

Antares Auto-Tune Pro 11
$459

PROS
- Remains an excellent pitch-processing tool.
- Useful streamlining of both Auto and Graph modes.
- The Harmony Player is an excellent addition.

CONS
- Triggering harmonies using MIDI notes would be a handy option.

SUMMARY

Auto-Tune has offered more than automatic pitch correction for some time but the new Harmony Player, which is easy to use and can produce very effective results, is a notable addition to its feature set.

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Sometimes, though, you need to get more surgical with your pitch correction, and for that there's Graph mode. This gives you full control over the pitch curve, pitch lines and note objects. Yes, such editing does require a little more time and effort from the user, but Graph mode offers a comprehensive toolset for corrective and creative editing of both pitch and timing within the performance.

Sweet Harmony?

Amongst a number of gentle cosmetic changes, the default size of v11's GUI is larger than before, and that's largely to accommodate some functional changes. For example, it enables the display of complete labels for all the controls. Antares have indicated that this move is also in response to user requests, but you can, of course, still resize the GUI to fit in the available screen space. The Preset Manager has also been revamped and new presets have been created, but this update is about more than cosmetic adjustments, and Auto mode has a number of new features.

First, if you are using the Transpose control (located towards the right end of the topmost set of controls) to apply a global pitch-shift to your vocal, you now also get the Conform To Scale button (a musical clef symbol). If you have set a specific key/scale combination, with Conform To Scale active, any Transpose setting is applied with that key/scale taken into account, so you avoid any out-of-key notes.

Second, mapping Auto-Tune's various controls or buttons to a MIDI controller is now easier, thanks to a MIDI Learn feature. If you right-click on any control, a pop-up menu presents the required options and you can either pick a suitable CC number, or 'learn' a controller automatically. This works well enough, though there are some limitations when using the VST3 version of the plug-in (as I was, within Cubase Pro 13 during testing), where the 'learn' option is not currently available. Hopefully, that's something that will be added in an update at some stage. MIDI Learn is fully supported within the AU version, though, and it worked fine when I briefly tested it using Logic Pro.

Third, we come to the undoubted headline feature of the Pro 11 release: Auto mode's new Harmony Player. This provides you with up to four harmony voices that are generated in real time from the vocal input (it can be used with non-vocal audio also!). If you've used Antares' long-standing Harmony Engine plug-in, the underlying concept (and, I assume, some of the technology) is similar, but Harmony Player is intended to be more interactive and streamlined, so that the harmonies can be triggered ('played') in real time.

To access this functionality, you just hit the Harmony Player button (top centre, just beneath the preset browser). This changes the central portion of the display, moving the standard Auto Mode controls to a smaller panel on the left. The rest of this central part of the display then provides controls for up to four harmony voices (in the middle) and envelope, EQ, gate and stereo width controls (on the right) that can be applied to those harmonies. To make the controls active, you also need to switch on the Harmony Player button (found in the topmost control strip, alongside the Modern/Classic toggle button). It's also worth noting that Harmony Player is only currently available if you have chosen a chromatic, major or minor scale type in the Key/Scale settings.

The central Harmony Player controls are also divided into three sub-sections. On the left, the Mixer button toggles the central portion between four large 'toggle' buttons and more detailed controls for each of the four harmony voices, with smaller trigger buttons. The Input buttons let you solo either the audio input (after pitch correction) or just the harmonies. Harmony Interval lets you choose between fixed or scale-based harmony intervals, with the latter being more musical if you want to use something other than unison or octave harmonies based on the key/scale you have set. Finally, you can flip the pan mode between standard L-R or stereo width options.

On the right side, the four knobs arranged vertically are global controls that adjust the degree of random variation between the harmony voices and the input voice. These enable you to create either very tight harmonies or go for a looser sound. Whatever their settings, the values of each parameter are randomised across any of the active harmonies to create a more natural result.

With the aforementioned Mixer button engaged, the central portion of the Harmony Player provides individual controls for each of the four voices. You can pick an interval, and adjust the formant, pan and level. Each voice also has a solo button. During playback, you can activate a harmony voice by pressing and holding the respective trigger button with your mouse. A global trigger button at the base

Pro 11's highlight addition is Auto mode's new Harmony Player.

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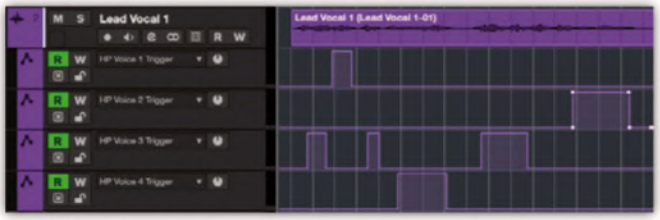
» of the panel lets you activate all four voices simultaneously. If you engage the padlock buttons, when clicked, the trigger switches stay on until clicked a second time.

Considering they're being generated in real time, I have to say that the quality of the harmony voices is good. Auditioned in isolation, you may hear some occasional artefacts that reveal the harmonies' artificial nature, but when placed in a mix the results are really rather impressive — up there with the best auto-harmony software I've used. But what I found truly — and surprisingly — impressive was just how creative the whole triggering approach could be in use. For example, having configured the four slots to provide an octave above, unison, an octave below and a third above (along with suitable formant, pan and level adjustments), the ability to then just 'play in' these harmonies during playback, harmonising with short phrases or even single words from a lead vocal, was very effective indeed. It's also a fun, not to mention very easy, way to create harmony parts, whether you intend to use the Auto-Tune audio directly, or get your singer to recreate the ideas you generated in this way.

Auto-Tune's controls are fully automatable so, in the context of a project, you can easily record (and edit if required) the mouse-based activation of each trigger pad, and this approach worked very smoothly for me in Cubase Pro 13. However, as far as I can see, Auto-Tune Pro 11 only responds to MIDI CCs — you can't, for example, activate the triggers using MIDI notes, and I reckon that would be a cool option to have compared with using a mouse to create automation data one trigger at a time, or the rather unwieldy option of turning four MIDI CC controllers through their full range to go from 'on' to 'off'. Some DAWs may offer workarounds for this (Cubase, for example, has the MIDI Input Transformer, with options to convert MIDI note data into MIDI CC data), but it would be nice to see it as a native feature of the Auto-Tune plug-in. That observation aside, the Harmony Player is a great new string to the Auto-Tune bow.



As shown here in Cubase, the Harmony Player's triggers can be automated easily.

Amongst other enhancements, Graph mode now includes a useful Zoom Outline function at the foot of the display.

Less Graft In Graph Mode

While the overall functionality remains the same as in Pro X, the Graph mode has been treated to some useful refinements. A 'helper' screen now pops open on first use, and this offers a quick reminder of the steps involved in getting your audio into the Graph mode editor. Of more long-term use are options to toggle on/off some toolbar elements — you can declutter the GUI if working on a more compact screen — and some new key command modifier options that allow you to quickly adjust the Per-Object Retune Speed (R), Vibrato (V) and Formant (F) controls while dragging or scrolling with the mouse. Also useful is the new Zoom Outline feature at the bottom of the Graph mode display. While it's a familiar concept that's found in many other audio editing environments, that doesn't stop it being very useful addition here for adjusting zoom and for navigation through your project.

Eleventh Heaven?

There has been more to Auto-Tune Pro than auto-tuning for quite some time, but in Pro 11 it's now also about vocal harmonies. But setting aside the new Harmony Player feature set for a moment, I think it's fair to describe the changes in this release as evolutionary rather than revolutionary. That's no bad thing in my view, and I'm sure existing users will welcome the continued improvements that allow a speedier workflow with less by way of significant disruption.

Auto-Tune Pro 11 is automatically available for Unlimited subscribers, but for those with a perpetual licence the key determining factor in an upgrade decision will be whether they can see themselves getting much use from the new Harmony Player. Personally, I think it's a great addition and I'm sure many users will see a benefit from it too, because as well as delivering instant harmonies it's a very creative tool for generating new harmony lines. Key to this is the ability to 'perform' with the triggers and record, and subsequently edit if needed, that performance as automation data. As I said above, being able to activate the triggers with MIDI notes would be a useful extra but, even so, this is a cool and genuinely useful tool.

For new prospective users, Auto-Tune Pro 11 should remain every bit as desirable as it always has been, and it's worth pointing out that while the purchase price remains the same as before you are now getting a better product for your money. As mentioned earlier, there are various alternative products for this range of pitch-based tasks, each with their own pros and cons. But for its core tasks of pitch correction, rewriting vocal melodies, creative pitch effects — and, now, generating vocal harmonies — Auto-Tune Pro 11 is an excellent choice. ■■■

$ Auto-Tune Pro 11 $459 (perpetual license, but includes a six-month subscription to Auto-Tune Unlimited; upgrades from earlier versions also available). Auto-Tune Unlimited subscriptions start at $14.58 per month.
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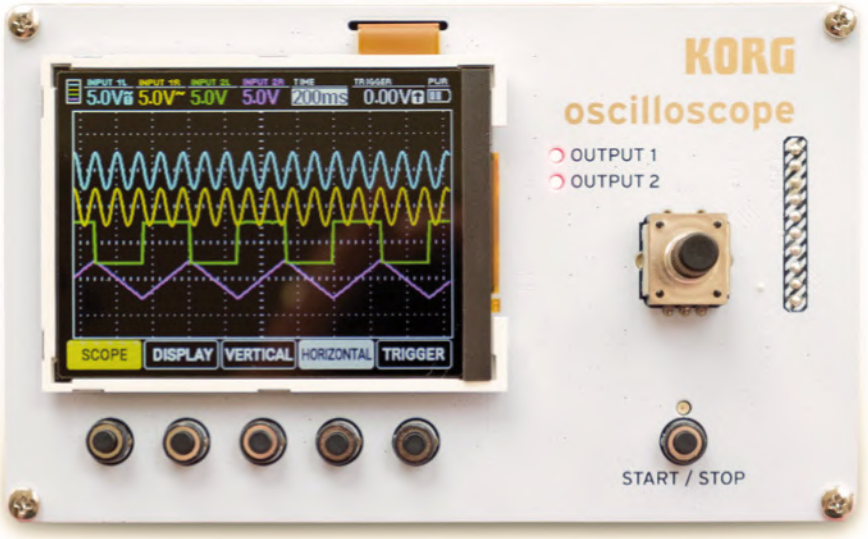
Korg NTS-2

Oscilloscope Kit

An oscilloscope can be very handy when working with synths and CV signals — but most aren't designed with music-makers in mind.



HUGH ROBJOHNS

Modular synthesizers are fashionable again and pleasingly affordable, and many conventional synths now include control voltage (CV) and gate connections for interoperability with modular hardware, drum machines, and even effects devices. When connecting these things together, it's often helpful if you can view the control voltage and gate signals to check whether they are actually doing what is expected, and it's often handy to view oscillator waveforms too. The device needed to do this is called an oscilloscope.

Traditional analogue oscilloscopes with cathode ray display tubes tend to be large, heavy, expensive and quite technical to configure and use, and they aren't equipped with audio-friendly connectors. Modern digital 'scopes aren't much better either. Added to which, an oscilloscope usually only does the one thing — it displays the waveform from one or two varying input signals.

Korg's Nu:Tekt NTS-2 oscilloscope kit is rather different, in that it's very much intended for use by music makers — in fact, Korg describe it as "the musician's Swiss Army knife"! This pocket-sized device features a 2.8-inch colour LCD screen and is less expensive than most proper 'scopes, yet provides four — yes four! — input channels, all of which have standard 3.5mm TRS jack sockets. More than that, though, the NTS-2 also provides an FFT mode that allows you to analyse the harmonic spectrum of input signals, plus two independent and very comprehensive signal generators (which are permanently available), and a very useful pitch tuner mode.

Construction

The NTS-2 comes as a complete kit of parts with a simple four-stage construction process that can be completed in about 15 minutes (an assembly tutorial video is available: https://youtu.be/Fx2wZ-Baz7g). There are two main circuit boards in protective bubble wrap, along with some aluminium corner pieces, a bag of screws, and a teeny-tiny screwdriver. There's also a 3.5mm TRS-TRS audio cable, a USB A-C cable for powering, and some stick-on rubber feet.

Assembly starts with a slightly nervewracking process of snapping small rear and front panels away from the bottom battery panel and top display PCB — the whole lot is manufactured as one large fibreglass piece, which presumably keeps costs down. But although some care and patience is required, it's not difficult. In the same way, a couple of small spacer pieces are snapped away from the lower input PCB, although these and two others are only required if you want to make and attach side brackets to support the display on a desktop (a template is included for making said DIY side panels).

The four aluminium corner extrusions are bolted onto a pair of aluminium side panels using tiny bolts, and then these completed side pieces are affixed with larger bolts to the bottom board carrying the battery holder. I found these small bolts a little fiddly to work with, and preferred using a jeweller's screwdriver over the miniature one included in the kit, but with patience and a steady hand I completed the task without any cursing.

Next, the input connector board is plugged into the back of the top display »

The NTS-2 comes as a self-assembly kit with pre-populated circuit boards — assembly took me only 15 minutes.

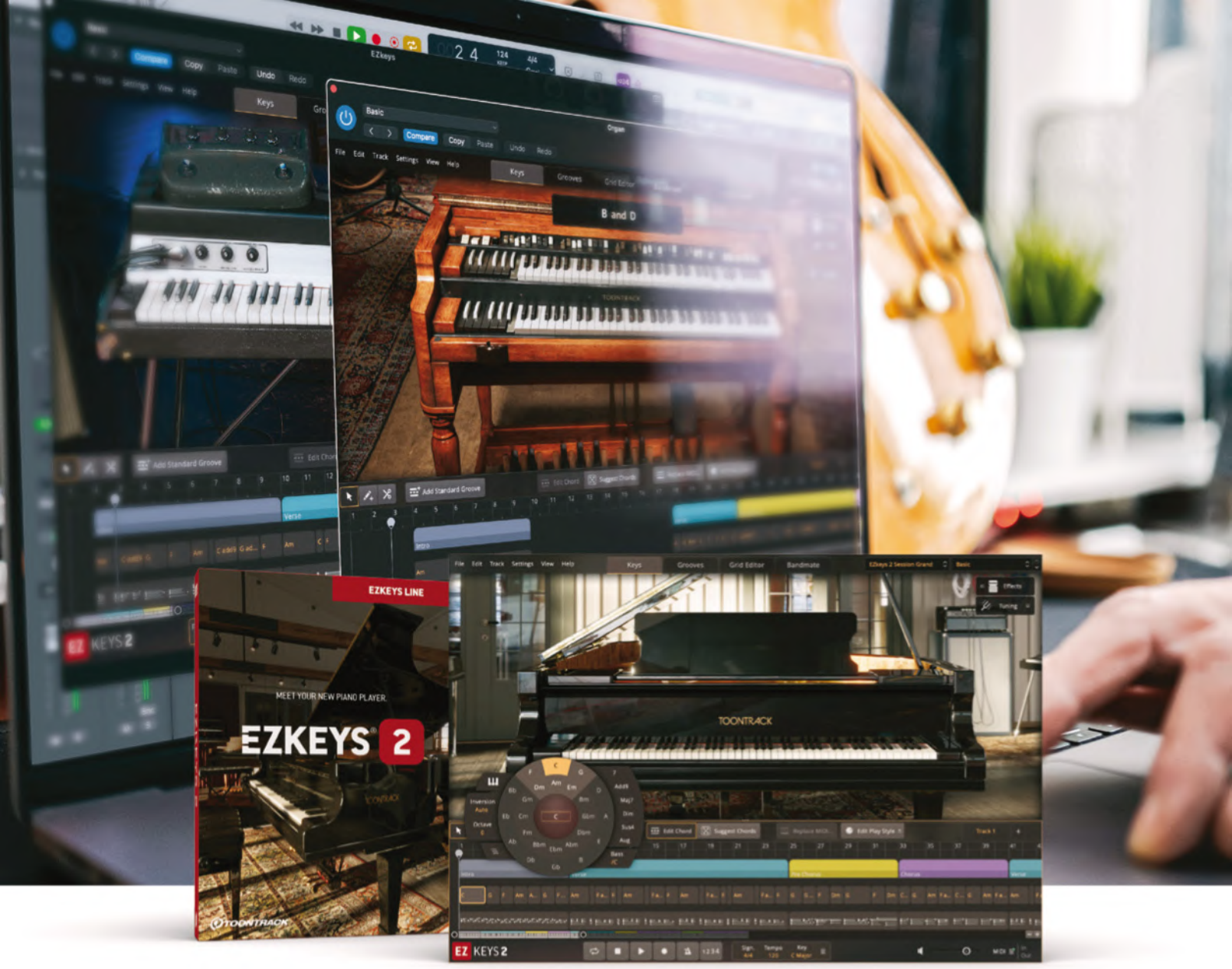
Korg NTS-2
$200

PROS
- Simple plug-and-screw together kit assembly.
- Four-channel scope.
- Fully optimised for Audio/ CV applications.
- Useful feature set includes FFT, Tuner and dual signal generators.
- Thru audio outputs allow permanent connection within a system.

CONS
- Physical user manual not included.
- Assembly is fiddly due to tiny screws.

SUMMARY

A skilfully optimised four-channel digital 'scope kit for simple DIY assembly, ideal for any modular synth setup, with spectrum analysis and tuner functions, as well as two versatile signal function generators.



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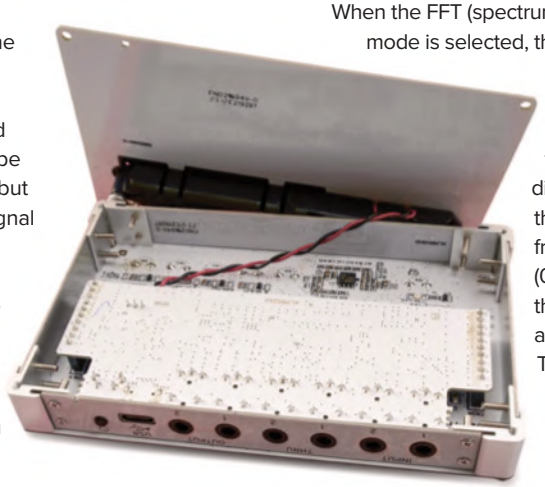
» board, and then the cable from the battery holder is plugged into a socket at the side of the input board. Finally, the whole top assembly can be screwed down onto the corner pieces and the front and back panels fitted with more tiny screws — the back panel carrying printed labels for all the connectors and on/off button. And that's it: assembly is complete, and the unit ready to use.

Ready, Set, Go

The completed Oscilloscope measures 129x78x39mm and weighs about 130g, so it's genuinely pocket-sized! Two AAA alkaline batteries will run the 'scope for a few hours, but I think most will use the included USB cable to derive power from either a standard charger wall-wart (providing 5V DC and at least 550mA) or a computer. Separate Auto-Off durations are available for battery and USB powered operation (the default is one hour for both, but there's a permanently on option, too).

The four input channels are accessed via a pair of 3.5mm TRS (stereo) sockets on the rear, so breakout Y-cables might be required in some applications (I used 3.5mm TRS to dual 6.35mm TS cables, as most of my gear uses quarter-inch CV/gate connections). If mono TS plugs are used only two channels can be accessed, of course. Usefully, both input sockets have associated (stereo) thru output sockets, so the NTS-2 can be patched permanently across up to four signal paths, if required. A third pair of 3.5mm sockets, monaural TS types, provides the outputs from the two internal signal generators. The last connector is a USB-C port for powering, while a power button turns the unit on/off.

Moving on to the top control panel, five buttons are arranged across the bottom of the LCD screen, the first choosing the working mode. The functions of the other four vary depending on the current display mode. A sixth button starts and stops oscilloscope measurements, but also turns the signal generator outputs on/off when configuring them (a pair of LEDs light when the generator outputs are active). A knobless rotary encoder has a detented action and alters the value of the currently selected on-screen parameter.



There are four inputs and outputs, with two channels on each of two minijack sockets, and there's a handy pair of thru sockets too.

The operating paradigm is obviously menu-driven and mostly self-explanatory, although I did have a few head-scratching moments (such as wondering why I couldn't get outputs from the generators!). The kit didn't include a printed user manual, but thankfully I was able to download one from the web (https://cdn.korg.com/us/support/download/files/f5ddb3476dc7b90ddac5fb1013428a27.pdf). It's only four pages long but I'd definitely recommend reading it to learn about all the features and operations the NTS-2 is capable of — this is a surprisingly powerful and capable product that really does justify the Swiss Army knife description!

Operations

Pressing the Mode button cycles around the four core functions of Oscilloscope, Signal Generator, Spectrum Analyzer and Tuner, with a fifth option for Global settings. In 'Scope mode, the assignable buttons access the display mode (overlaid, two- or four-channel displays, and X/Y Lissajous), vertical and horizontal resolutions (10mV to 10V, and 50µs to 1 second per division), and the triggering mode and sensitivity.

The Wave mode accesses the two independent signal generators, each working as an audio oscillator (sine, square, triangle, or saw wave shapes), an LFO, a noise generator (pink or white), and a pulse/envelope generator. Options are included to adjust the pulse/envelope bpm (0.5-600), as well as polarity, duty cycle and phase of the oscillator/LFO signals (both spanning 0.01Hz to 10kHz).

When the FFT (spectrum analyser) mode is selected, the configuration options include selecting the input source(s), the vertical display resolution, the horizontal frequency range (0-20 kHz), and the trigger level and type. In Tuner mode the reference pitch is adjustable between 410 and 480 Hz, any of the input sources can be selected individually or in pairs, the display type can be chosen (needle or linear meter), and the oscilloscope traces can even be displayed underneath, if required. The detected pitch is given in equal temperament notes between E0 and G8, accurate to ±0.1 cents.

Global settings include things like AC or DC coupling, low-pass filtering (to remove HF noise and clean up the traces), and the trigger source for the 'scope inputs. Display brightness, auto-off durations, and a couple of other display options are also provided. For technical completeness, the input impedance is a whopping 1MΩ on all channels, with ±10V (+19dBu) maximum input level, while the generator outputs are limited to a ±5V range (+13dBu) with a low 47Ω output impedance.

Impressions

Korg's Nu:Tekt NTS-2 kit is impressive, and very satisfying to both assemble and use. I have already found it invaluable for checking control voltages, gates and audio waveforms in my modest synth rig, which comprises a Moog Sequential 37CV, an ARP Odyssey, a Korg MS20FS, a CP251 controller, and various MoogerFooger effects units.

Undoubtedly, there are cheaper and simpler digital LCD oscilloscopes on the market, as well as many pre-loved vintage CRT units, but the NTS-2 has the real-world advantage of being fully optimised for exploring audio and CV signals, with sensible control ranges and familiar connectors. The inclusion of an X/Y function in the 'scope makes for mesmerising displays, while the FFT mode is really welcome for analysing the harmonic content of audio signals. Likewise, the tuner function gives reassurance and helps align synth oscillators to a known reference. The two independent signal generators are also extremely useful not only for system testing, but also as additional audio, LFO or trigger sources within any synth setup.

I can see this oscilloscope kit being a very useful tool in any modular or semi-modular rig, and it lends itself to being built into a custom rack panel, console frame, or used as it comes, as a free-standing desktop unit. Either way, it always looks very cool while providing really useful information. I'm very glad I bought one — I use it almost every day!

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Heritage Audio RAM 1000

Passive Monitor Controller

The RAM 1000 combines hands-on speaker control with Bluetooth streaming.

MATT HOUGHTON

In these post mixing-console days, I reckon we could all use a good monitor controller: hands-on control of the levels in your studio, whether the computer is switched on or not, is a good thing, as is an obvious analogue mute button that you can hit in the event of horrible noises emanating from said computer! Billed as "the solution for mid-sized studios", Heritage Audio's RAM 1000 is a desktop-format passive monitor controller that can cater for two stereo analogue inputs, two stereo analogue outputs as well as something which is less commonly seen on devices such as this: a stereo Bluetooth input.

Layout & Construction

The RAM 1000 is rather chunkier in the flesh than it seems in context-free online images, and it has both rubber feet and reassuring heft: even though the input and output cables trail from the rear (as they do with all such desktop designs) those cables certainly won't be dragging the RAM around your desk. The folded-steel case forms a box with a sloped top panel whose top and bottom edges curve pleasingly, with a nice finish and no sharp corners or jagged edges.

Around the back are two pairs of stereo analogue inputs, one on Zwee-branded combi XLRs, the other on TRS jacks, and the latter has a switch for +4dBu or -10dBv operation. When nothing's plugged into the right-channel jack, the left-channel jack doubles up as a mono input. Opposite are the two stereo output pairs, one on dual male XLRs (this one also has a Rec/Mon switch) and the other, sans switch, on TRS jacks. Between the inputs and outputs you'll find a single quarter-inch headphone output (might it have been better placed on the front panel?) and a grounding point. There's also a USB-C power socket, because although the level control itself is passive, power is required for other functions, including a helpful input level meter. A short USB-C to USB-C cable is supplied. There's no PSU but it's not a 'thirsty' device, and it ran very happily off a MacBook Pro battery for quite some time.

All the analogue I/O are on the rear panel, with both XLR and TRS jack options available.

On the top panel, the controls all feel reassuringly robust. There are switches for each input and output, to dim (-20dB) or mute the output, and for a mono check, and these are all latching types. If multiple sources are selected simultaneously, they're mixed, so all play through the selected outputs. I never had an issue with any of the buttons, but sometimes I did wish they had indicator LEDs, since it's not always easy to see what's selected at a glance, particularly in an 'atmospherically' lit studio.

The master level knob is a joy: it's big and easy to turn and, since it operates a switch with 23 steps of 3dB rather than

Heritage Audio
RAM 1000
$499

PROS
- Stepped master level control for accuracy and recall.
- Sounds transparent.
- Can mix (not just switch) inputs.
- Bluetooth input is so convenient.
- Solid construction and neat build quality.
- USB power socket.

CONS
- Some may find it light on facilities.

SUMMARY
A high-quality two-in, two-out monitor controller, with a very handy bonus: a third stereo input via Bluetooth.

a pot, there's good level matching between the left and right channels right through the range. This means it's really easy to set and recall reference levels too: the steps are obvious, and the size of the knob means there's an obvious physical distance between each one — it's pretty hard to dial in the wrong setting by mistake. The headphone level knob also has plenty of range, and the amp drove my different studio headphones of various impedances effortlessly. My big sausage fingers would have preferred that this knob had greater girth, though, or perhaps a rubberised finish to aid grip. Still, it works and, once again, it's easy to recall settings, particularly if you were to mark reference levels with a DIY label or chinagraph pencil. The only other control is associated with the Bluetooth input: a momentary button to reset the Bluetooth connection. Speaking of which, several Bluetooth codecs are catered for, to allow the best possible Bluetooth audio quality: it automatically matches the streaming device's settings.

What's Not To Like?

The RAM 1000 does what it says on the tin, and it does so very nicely too. It sounds good, with zero audible noise or distortion to my ears. The layout is intuitive, switching and recalling settings is easy as pie, and setting up a Bluetooth connection was easy, so if an artist brings a demo to your studio or you want to ask about references, they can play tracks from their phone over your studio monitors without fuss. It's very handy that you can dedicate one output to the monitors and another to a recording device. I also like that power is over USB — you shouldn't have any problem sourcing a replacement supply in the future or when on location, should you need to!

While the dim, mute and mono buttons all serve useful purposes, though, and that big master level switch is pleasing, I generally prefer to have more facilities available on a monitor controller and personally I missed a few things here. For example, I like to be able to mute the left and right output channels independently, and ideally to audition the Sides-only signal (eg. by polarity inverting one channel). It's possible to do that in your DAW or another device hooked up to the RAM, but the whole point of a monitor controller is to make this sort of thing convenient. Also, I did sometimes wish that I could mute the headphone output, so that I could leave open-backed cans attached without hearing their spill, and without changing the level setting. There's no dedicated provision for single speakers either, such as a subwoofer or Auratone-style grotbox. To be fair to Heritage, though, they offer a range of monitor controllers, and some have more channels and functions. On the whole, if you're happy with the facilities on offer here — and I've no doubt that many will find them sufficient — then the RAM 1000 is a great-sounding, robustly built monitor controller that could prove a great choice and seems like it would last a lifetime. The competition in this market has heated up in recent years, with good options available from the likes of Drawmer and Audient amongst others, but while you should always weigh up your options, I'd have no hesitation in recommending the RAM 1000.

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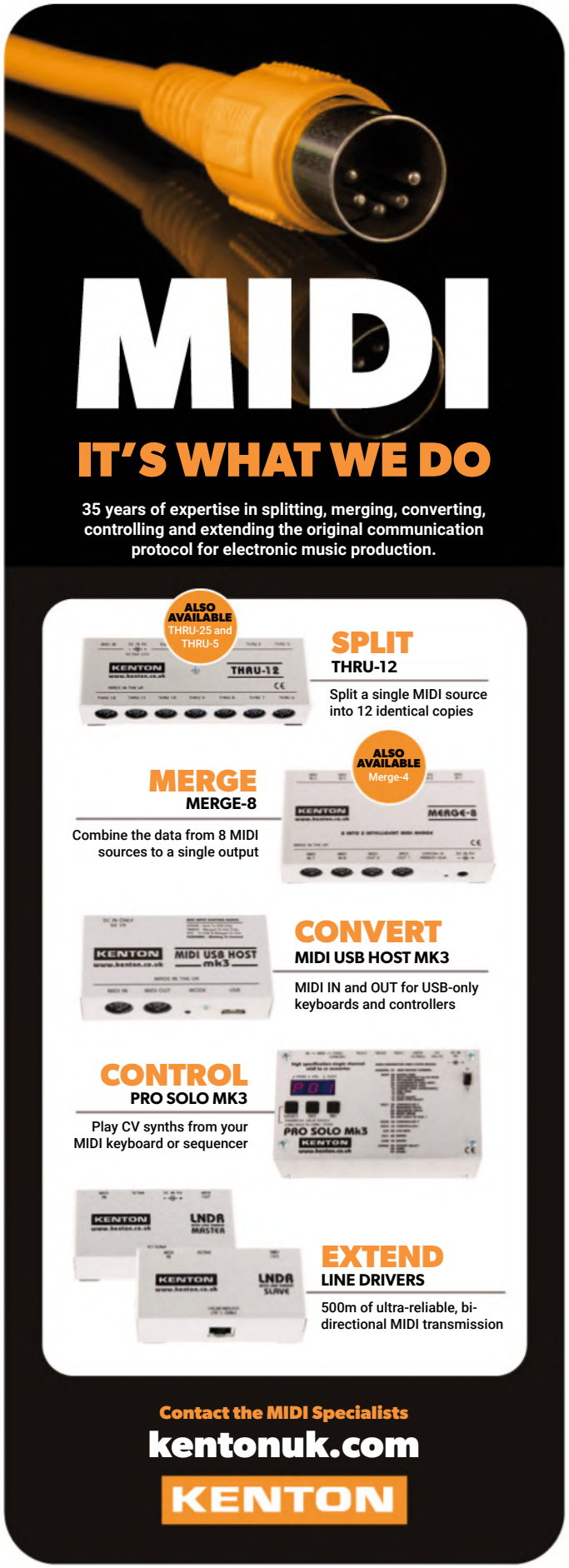

Drawmer Electronics 1977

Analogue Channel Strip

Drawmer have been offering no-nonsense professional audio gear since the early ’80s, but their current range of processors offers more possibilities than ever.

MATT HOUGHTON

British company Drawmer Electronics have been around since 1982 and today they offer several lines of outboard processors, along with monitor controllers, master clocks, signal routers and even plug-ins, via a collaboration with Softube. For review here is their latest analogue channel strip: the 1977. It’s a mono, 2U rackmount device that combines a preamplifier, an EQ, a FET compressor and a saturator with some useful routing facilities. It connects to mains AC via an IEC socket on the rear, and can operate on 230V 50Hz or 115V 60Hz power — to prevent accidents you must unscrew a metal panel and insert a suitable fuse if you wish to change this.

Externally, the build quality is good: unfussy, solid and fit for purpose, with very clear panel markings. On the inside most components are traditional through-hole types, laid out on one main motherboard that fills the chassis, and a vertical daughterboard behind the front panel.

Preamp & EQ

The preamp has separate mic and line inputs on the rear (electronically balanced XLRs), along with a TS jack instrument input on the front. A front-panel rotary switch selects the source, and there’s 0 to 66 dB of gain available for mics and -24 to 42 dB for line or instrument sources. For the mic input you can choose between four positions on the selector switch. The first three offer different input impedances (2.4kΩ, 600Ω and 200Ω) and no phantom power — these can be useful when working with passive dynamic (moving-coil or ribbon) mics, as the impedance can have a significant impact on their tonality. This arrangement also ensures phantom isn’t present where it’s not wanted. The fourth position is for capacitor (and other active) mics and this engages 48V phantom power automatically.

There are some more typical preamp features here: a polarity inverter is joined by a high-pass filter; helpfully, the latter is fully variable from 16 to 130 Hz, making it suitable for use as an unobtrusive rumble filter, to counteract the tonal impact of the proximity effect or, for example, to roll off the bottom end of a guitar to prevent it treading on the toes of the bass. This filter can also be switched in/out of the signal path without changing its frequency setting.

Conveniently, alongside the instrument jack another quarter-inch TS jack provides a ‘thru’ output, and a similarly thoughtful touch is the preamp direct out on another electronically balanced XLR on the rear. So you can use the instrument stage like a traditional DI box, with the preamp’s line output going to FOH or a recording device and the thru feeding your amp or pedalboard. Or you might record the clean signal post the preamp and use the processing stages later when mixing, or even at the same time when streaming or broadcasting.

After the preamp, there are several processing stages, including a three-band equaliser. This can be switched in or

out of the signal path (it's a hard-wired bypass) and pre/post the compressor, courtesy of a button to the left of the EQ section. A ±12dB low shelf can be set anywhere from 40 to 725 Hz, and a switch allows you to choose 6, 9 or 12 dB/octave slopes. There's also a really useful Peak setting — this once again has a 12dB/octave slope but this time there's a resonant peak around the turnover frequency. The mid band, a parametric bell, has a generous 140Hz to 13kHz range, and bandwidth (0.25 to 2.75 octaves) and ±12dB gain controls. The high band, another ±12dB shelf, can be set from 530Hz to 20kHz and there are 6 and 12 dB/octave slopes available.

Compressor & Saturator

The FET compressor has a threshold knob (+20 to -40 dB) and, alongside more conventional controls, there are buttons marked Big and Air. I'd initially assumed that Big might be some sort of parallel compression setting, but in fact it engages a high-pass filter in the side-chain, thus reducing the compressor's sensitivity to low frequencies, which often leaves the bottom end sounding tighter and more solid. So it's a useful feature, particularly if using a pair of these on a drum or mix bus. But should you wish to do more to influence the compressor's response, there's an unbalanced TRS side-chain insert point on the rear. I tend to prefer this approach to a simple key input, as you can press into service even the cheapest, nastiest sounding EQ to help you get more from the compressor! But I was pleased to discover that, via a Y-lead from my patchbay, I could also use the side-chain insert's return as a key input for side-chain compression, for example using a kick to duck a bass guitar. Air, meanwhile, seems to be an HF EQ cut, to prevent the compressor clamping down too hard on sounds with a lot of their energy in the top end.

There are conventional controls for the ratio, attack and release. The ratio runs from 1:1 (no compression) to 10:1 (effectively limiting), while a continuously variable attack control ranges from 0.2 to 100 ms and a similar release control from 0.05 to 2 seconds. A separate button, PGM, engages a programme-dependent (or 'auto') release mode (for more on what an auto release does check out Hugh Robjohns' explanation in *SOS* September 2014: www.soundonsound.com/sound-advice/q-what-does-compressors-auto-release-control-do).

Before we get to the make-up gain knob (-10 to +20 dB), there's a saturation processor, comprising a single knob and two buttons, one to engage a high-pass and the other a low-pass filter. The knob is marked simply 1-10, and you just turn it clockwise for more distortion. The filters, whose frequencies aren't specified, help you to tame any ill effects heard in the high and low end, focusing the added energy more in the mids when both are used — that's often where you'll want it, and I found both essential when using saturation on vocals in particular. After the make-up gain, a Comp In button switches the whole compressor-plus-saturation section in/out of circuit, so if you set the make-up gain appropriately you can easily A/B compare the compressed and uncompressed signals.

There's also a small 'master section' on the right, comprising an output level knob labelled Fade (full attenuation to +12dB of gain) and, just before this in the signal path, a wet/dry mix pot. The latter applies to the whole signal chain post »

Drawmer Electronics
1977
$1799

PROS
- Clean and versatile preamp stage.
- Generous processing options.
- Compressor side-chain insert point.
- Keen price.
- Sounds great.

CONS
- Nothing of note.

SUMMARY
A lovely sounding and wonderfully versatile recording channel.

» the preamp, which means you can set the balance between the dry sound from the preamp and the entire processing chain (EQ, compressor, saturation). It's common to have a wet/dry mix control on compressors these days, but I see it much less often on full channel strips, and it's a real bonus here because, with the EQ pre/post-compressor switching, it means you're able to shape the compressed path in the way many engineers like to when using parallel processing on a console. The same goes for parallel saturation: while you have its built-in filters already, you could effectively bypass the compressor by setting the threshold high enough that there's no gain reduction, then just have parallel saturation, with the saturation path shaped by the onboard EQ section. It's this sort of thing that makes me think of the 1977 as a Swiss Army processor — it's not likely to lay idle in your rack!

Impressions

Features and possibilities are all well and good, but that's all for nothing if a device doesn't deliver the goods sonically. Happily, the 1977 does! It's intended for a range of sources, from vocals and acoustic instruments, to electric guitars and synths, and even (used in pairs) the drum and mix busses. With other processing stages on board, then, it's not surprising that the preamp seems to have been designed with cleanliness in mind: it works very well to bring signals up to the desired level: clean, detailed and quiet, just as you'd want. When pushed too hard there's a rapid transition to breakup, so I wouldn't be looking to drive this preamp into a sweet spot. But there's a saturation stage for that, and also potentially some mic-dependent colour courtesy of the input impedance options.

Metering

In between each pair of EQ bands is an overload LED that illuminates when the EQ stage is overloading — an indication to dial back the EQ or preamp gain controls. For the compressor, gain reduction is indicated by an eight-segment (1, 2, 3, 5, 7, 10, 15 and 20 dB) LED bar-graph meter and, on the right of the unit, there's a large backlit moving-coil VU meter that displays the output level. As shipped, 0VU equates to +4dBu, and to calibrate it for a different operating level there's a preset on the main circuit board. There's also a 10dB pad switch on the front, to make the meter more meaningful when working with deliberately 'hot' signals. Since each processing section has a hard bypass, and another switch on the far right toggles between the post-preamp and post-processing signals, it's easy to establish the level at different points in the chain. Another neat feature is that you're treated to a light show, whereby the edges of the VU meter flash red as the saturation circuitry does its thing. I like this, though was surprised that it didn't cease when the compressor/saturation stage was bypassed, and some might prefer the ability to turn it off.

The EQ strikes me as a good, versatile console-style EQ. Not only is there considerable overlap in the ranges, meaning you can, for example, boost the low end of a kick using a shelf and then apply a narrow parametric cut in that 200Hz region that can sound boxy. But the shelves also offer way more control than you'd suspect. Sticking with the kick example, with the low band's peak setting engaged, you get both a resonant peak around the centre frequency and a corresponding narrow dip just above it — so you can emphasise a note in a drum without too much collateral damage. And if doing that raises too much flabby low end, the variable high-pass filter in the preamp stage can compensate for that. Subjectively, it sounds sweet too: it's easy to get it to do what you want, and never sounds harsh or scratchy.

The compressor seems to have a soft knee, and in general with lower ratios it sounds smooth and graceful. That makes it really nice on vocals, amongst other things, but I could also use more assertive settings to really pummel a (mono) drum bus, a role in which the Big side-chain filter helps achieve a solid, powerful low end, and the wet/dry blend control can be used to restore some dynamics. No complaints there! It's the saturation control, though, that makes this whole section rather more special. While very audible, it's not an in-your-face distortion (it sounds classier than that) and it worked well on most sources that can use a little grit and dirt. But in that parallel drum compressor role it excelled, doing some lovely things to the snare in particular.

In summary, Drawmer have combined elements of their 1970s range in one package, and the result is a very well specified channel strip that delivers the goods sonically. Almost everything you could want in a channel is here, the one exception perhaps being a de-esser, and it could easily serve as a studio's main recording channel: a vocal strip, a front end for your guitar, bass or synths and more, and still offering plenty of options at mixdown. The build quality is decent, there are nicely wide-ranging controls, and there's lots of configurability. Well done, Drawmer! ■■■

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Focal ST6 Trio6

Active Monitors

Single-driver, two-way nearfield or three-way monitors? Focal's new flagship design can be all three...

PAUL WHITE

Focal have a strong track record in producing serious studio monitors, and while some of their models are on the costly side for the home studio operator, they are actually very competitively priced in professional terms.

The ST6 Trio 6 is the latest in the ST6 series, and while you might assume that the Trio in its name relates to the number of drivers (there are indeed three), it in fact refers to the fact that the speaker can play three different roles, all of which are valuable in today's studios. This is down to what Focal call Focus mode. It's a feature we have seen before in earlier models, but whereas those offered two operating modes, the ST6 Trio 6 offers three.

The ST6 Trio 6 is a three-way speaker, then, incorporating an 8-inch woofer for the bass, a 5-inch driver for the mids and a 1.5-inch tweeter for the highs. Perhaps unusually for a modern premium monitor, its signal path is all analogue.

Theory Of Evolution

Like other models in the ST6 range, the new Trio 6 is an evolution of the earlier 'Be' version, where Be denoted the use of a beryllium tweeter. Curiously, the beryllium tweeter remains, but the 'Be' in the product name doesn't. However, the ST6 series includes a number of other technical improvements to the components, such as the use of Focal's W composite woofer cone construction, which features layers of woven glass-fibre composite around a structural foam inner core. The midrange driver has also been tweaked so that it now includes Focal's distinctive ridged roll surround, with the aim of further minimising resonances — Focal call this their Tuned Mass Damper resonance-reduction technology. We're informed that the radial rings moulded into the surround behave as tuned mass dampers to create anti-resonance nodes that dissipate energy rather than allowing it to reflect back into the diaphragm. And although the beryllium tweeter diaphragm itself is the same, it gains a newly designed asymmetric waveguide to control its dispersion.

There's plenty of amplifier power on board, with a 100 Watt Class-G amplifier driving the bass end and another handling the mids. Class-G amplifiers are analogue and utilise multiple power rails at different voltages. They switch from lower to higher power rails as the signal output approaches

Focal ST6 Trio6
$3499

PROS
- Accurate, open and detailed sound.
- Excellent stereo imaging.
- Focus modes might be useful to some.

CONS
- None.

SUMMARY
In their full-fat three-way mode these are serious, professional tools that will let you know exactly what your audio is doing. The Focus modes increase their flexibility, but sound rather better than the 'grotboxes' they try to emulate!

the maximum that each rail can handle, which in turn reduces power wastage in the output transistors. A 50W Class-A/B amplifier drives the tweeter. Maximum SPL is quoted as 113dB in full-range mode and 112dB in Focus mode.

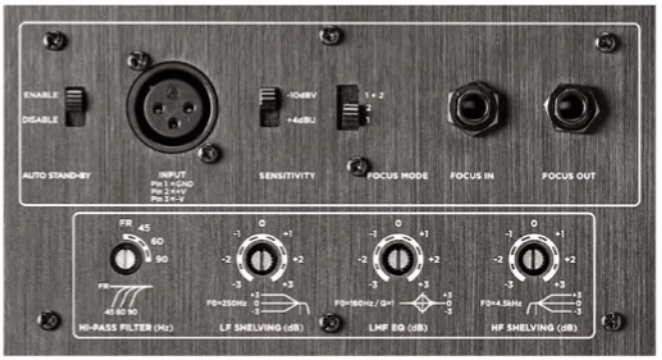
The beautifully styled cabinet housing these components has also been re-engineered, with upgraded bracing to further minimise resonances. With each speaker weighing around 20kg, you may need help to lift them onto your meter bridge. Internally the cabinet is divided into separate compartments for each driver, and the tweeter enclosure is apparently designed to behave as though it contains an infinite volume of air, in order to optimise acoustic loading. Unusually, both the mid and bass driver enclosures are ported, presumably so that in two-way mode, the midrange driver can behave in the same way as the woofer in a smaller ported system. Both sets of ports are flared not only externally but also internally where they terminate inside the cabinet, to encourage non-turbulent airflow, which reduces port noise at higher playback volumes.

Conscious of the fact that some studio layouts make it more practical to use monitors in landscape rather than portrait orientation, Focal have engineered the baffle such that the tweeter and midrange driver share a rotatable faceplate to optimise performance in either role. Rotation is achieved by removing the four cap-head bolts with an Allen key, rotating the plate by 90 degrees, then replacing the bolts.

In Focus

Returning to Focus mode, this reconfigures the drivers and crossovers from their usual three-way mode of working to provide two alternative monitoring options, using either one or two of the drivers.

Previous Focus-enabled monitors from Focal have allowed the tweeter, woofer and crossovers to be temporarily reconfigured so that the midrange driver handles the full audio range, rather like the revered Auratone. That mode is still available, but Focal have added a third mode that utilises the midrange driver and the tweeter but bypasses the woofer, to approximate the characteristics of a smaller two-way speaker such as those used in some hi-fi systems or small studio monitors — though the fidelity on offer here is rather higher than from typical home hi-fi speakers! Likewise, the single-driver mode sounds better than I remember my old Auratones sounding, though you do lose high-frequency detail in a rather similar way. That said, these additions mean that more mix checks can be made without switching to alternate monitors; if you find yourself doing this a lot and want to make changing modes straightforward, a footswitch can be connected via a jack to one monitor, with another jack cable to link the two speakers so that they both switch modes together.



The rear panel houses the analogue inputs, EQ controls, and the switches and footswitch jacks for engaging the Focus modes.

Controls

The input sensitivity can be switched between +4dBu and -10dBV, and to help tune the speakers to different rooms and proximity to boundaries, there's a useful amount of EQ. An HF shelving filter works above 4.5kHz over a ±3dB range, while a LMF EQ provides adjustment around 160Hz with a Q=1 to help compensate for desk mounting. LF shelving offers an adjustment range of ±3dB below 150Hz. The power indicator LED shows green when the speakers are operating and red when in power saving mode. Power saving can be turned on or off using a rear-panel switch.

Although the Focus mode can be turned on and off remotely, the choice of Focus mode is set using a three-way switch on the rear panel. In the first mode, which uses only the mid driver, the frequency response is 100Hz-15kHz. The second Focus mode is the two-way mid/tweeter arrangement, with a much extended frequency range of 80Hz-40kHz. When Focus 1 mode is activate, the 'Focus' LED on the front panel lights up green. In Focus 2 mode it lights up orange. The 1+2 Focus switch position allows for remote selection of Focus mode one or two using a Boss FS-6 double control pedal or equivalent (this requires connection using a TRS jack). Alternatively, you could use two single footswitches to do the same job.

Testing

Set up on my studio desk bridge shelf, the Trio 6s sounded impressive on a wide range of material before I even considered adjusting the EQ or indeed giving the drivers time to run in. Loudspeakers do take time to run in, as the material properties of the drivers, particularly the surrounds, change slightly after being flexed in normal use; you can either wait for days for that to happen naturally as you use the speakers, or you can play some loud dance music through them for a day and hope the neighbours don't complain!

As with other really high-quality speakers, the Trio 6s exhibit a tight, well-controlled low end, with a sense of openness and detail extending right up the audio spectrum. They deliver that sense of three-dimensional sound and transparency that lets you hear right into a mix but without adding harshness. The tweeter extends an octave higher than most, and if you find its detailed sound a hint too forward in character, a small tweak to the rear-panel EQ brings it back into line. Their sense of stereo imaging is also solid. For my monitoring location I'd probably drop the tweeter level by a dB but that's about it. However, it may well be beneficial to use serious speaker platforms rather than the small stick-on feet that come with the monitors.

Overall I couldn't fault these speakers in their usual full-range mode where their format makes them suitable for use in small-to-medium sized professional studios as well as being a good fit for serious project studios. The alternative modes work as expected, though I'm not sure how useful they are, as the results sound simply too good in comparison with Auratone cubes or domestic two-way speakers! Nevertheless, the facility is there if you need it.

In summary, then, these are extremely fine speakers that fulfil the need for accurate monitoring and give you the two Focus modes as a bonus, whether or not not you decide to use them.

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Electra One MkII

MIDI Controller

Does a modern hardware MIDI controller with the ability to add your own features sound too good to be true?

RORY DOW

The Electra One, by a company of the same name, is a small desktop MIDI controller with a touchscreen, a dozen high-resolution rotary pots, six buttons, and USB and DIN MIDI ports. Its superpower is a mature Web MIDI platform that allows you to create custom controllers and switch between them at the touch of a button. There's even a scripting language enabling anyone with a small amount of programming or scripting skill to create custom features, including two-way MIDI communication, bulk SysEx interpretation, custom display formatting, and MIDI effects like arpeggiators and LFOs.

In essence, the Electra One can be a controller for DAWs, live performances or an editor/librarian for any MIDI equipment, new or old. At first, I was most excited about making editors for some older rack equipment in my studio, like the Roland MKS-50, Yamaha TX81Z, and the Korg 03R/W. These are great synths hamstrung by awful UX design, and many software editors no longer work on modern Macs and PCs. The idea of having a small desktop device that can easily edit sounds on these old synths, switching from one profile to another, with the added reliability of hardware, is very appealing. However, I quickly found other uses for the Electra One as a DAW controller and partner for specific pieces of equipment, essentially adding more physical controls and features.

MIDI Chameleon

At 206 x 189 x 38mm, the Electra One will take up very little space on your desktop. The aluminium enclosure is reassuringly solid, and there is no wobble in the touch-sensitive knobs. The screen is a 7-inch colour LCD with 1024x600 resolution and looks great. Power comes from the USB Type-B port on the back, which also supplies two virtual MIDI ports and a control port for sending data back and forth from the Web MIDI interface. While I would usually moan about the lack of a dedicated power supply, in this case, it's OK. The Electra One does not generate audio, so those pesky ground-loop hums caused by USB-powered equipment are not an issue.

On the rear of the unit, we also find a USB Type-A host port and two sets of MIDI DIN in/out ports. The USB host can be used to connect any USB MIDI class-compliant device. If you want to connect more devices, you can use a USB hub. All these MIDI ports can send and receive MIDI, and a programable MIDI router and merger allows you to configure the unit to your setup.

A typical setup might be USB Type-B to the computer for power, Web MIDI editing, and two-way DAW control; USB Type-A host port connected to a a master keyboard and modern sequencer for live performance enhancement; and MIDI DIN ports used for older equipment without USB. With this setup correctly configured, you can switch between presets to use the Electra One as a DAW controller, sequencer partner, or ancient synth editor without rewiring. You can also use the Web MIDI

environment any time to edit your presets and upload them to the unit.

Hands On

The Electra One screen can contain up to 36 parameters, and an instrument (or 'preset' in Electra One lingo) can have up to 12 pages, allowing you to configure up to 432 parameters per preset. The screen is divided into three rows with 12 parameters on each row. One row is active at a time, and the 12 endless encoders control the parameters of that row. You can use the buttons on the left to move between rows or just touch the screen.

A preset contains everything you need for one configuration — perhaps a synth, DAW, or something more unusual like a bank of MIDI LFOs or a custom MIDI Clock generator. If required, you can address up to 32 different devices from a single preset. Devices are configured separately and contain information about what MIDI channel they respond to and what port they are on. The unit can store 72 presets and switch between them at the touch of a button. On the newer MkII unit, an upgraded CPU allows you to run up to 12 presets concurrently, which means that even background presets continue to process. The older MkI units could only process one preset at a time. Once you transfer a preset to the unit, you can disconnect it from the computer and use it standalone — ideal for live performers wanting to take it on stage.

Creating your own presets is entirely optional. Electra One have a busy community of creators uploading their work to the Electra One preset library, part of the broader set of Web MIDI tools that communicate directly with the Electra One hardware. Adding one of these presets to your preset library and onwards to one of the 72 preset slots on the hardware is easily achieved using the Web MIDI online interface.

The library contains hundreds of presets, including synth editors, MIDI processors, sequencers, DAW controllers, and useful code examples. Sometimes, when using presets created by others, they need some tweaking to get working. MIDI can be a complex beast, and if the creator has set up their preset to use specific MIDI channels or SysEx IDs, you might have to edit the preset before it will work in your setup. If the preset has been configured correctly, this is trivial. However, sometimes, the preset creator may have written scripts that use specific MIDI channels, which can be more challenging to fix. As a non-programmer myself (although I dabble in a bit of scripting), I got most presets working, if not instantly then with a small amount of editing. And if you do get stuck, Electra One's helpful forum community is willing to help.

At the back of the Electra One we find a dual pair of MIDI I/O sockets and a brace of USB ports.

DIY

Although you don't need to, if you buy an Electra One, it's probably because you want to make something specific to your needs. The suite of tools available to assist you in this goal is impressive and, perhaps, a little bit daunting. The good news is that you don't need to do any scripting to get started.

The online preset editor works via Web MIDI, a browser API that allows your Web browser to communicate via your computer's MIDI ports. Anyone can create an Electra One MIDI controller by dragging and dropping controls in the editor. The Web editor mimics the hardware screen, and you drag and drop controls to slots and configure them to your needs.

There are five control types: fader, list, button, envelope, and 'relative' (a special control type for relative MIDI messages). Each control can be configured to send a MIDI message: NRPN, PRN, SysEx, aftertouch (poly or channel), pitch-bend, Program Change, and Continuous Controller (normal or relative). You can colour and name each control and change pages to fill more slots. Having named pages makes things easier to organise. If you are creating a controller for a synthesizer, you might have a page for oscillators, another for filters, and one for effects. You can also group parameters within a page so that controls are boxed off with headings like 'Oscillator 1', 'LFO 2', etc.

There is a lot of flexibility in individual controls. For example, let's say you create a control for a synthesizer's oscillator waveform parameter. You pick a fader that can scroll through the waveforms available. But instead of simply showing a number, you can specify text overlays for every value, allowing you to type in the name of each waveform and have those appear on the screen instead. You can even draw icons for each item using the built-in 48x18 pixel image editor, giving you a customised graphic for each waveform.

The envelope control type I mentioned earlier is a special case. It takes up a single slot on the screen but allows you to specify all the MIDI necessary to edit an ADSR, ADR or ADDSR envelope. Touching the slot on the screen opens a graphical envelope where you can drag the envelope stages, making editing intuitive and fast. »

Electra One MkII
€553

PROS

- **An incredibly flexible MIDI controller in a small footprint.**
- **Mature development environment with custom API and LUA scripting.**
- **A growing library of pre-made controllers available for download.**
- **Creating custom MIDI controllers doesn't have to be complicated but...**

CONS

- **...there is a steep learning curve if you want to make the most out of it.**

SUMMARY

The Electra One is a customisable MIDI controller based on a large touchscreen and 12 touch-sensitive encoders. Its ability to integrate new features via an impressive API and LUA scripting might just make it the most flexible hardware MIDI controller on the market.

» It's a shame only three envelope types are available, as many synthesizers and samplers have unique envelope designs. In these cases, you need to use a slot per envelope stage.

The Web MIDI preset editor has plenty of helpful tools. There's undo and redo. You can easily copy and paste controls and test your creations by monitoring their output in the MIDI monitor. You can even save revisions of your preset and revert to a previous revision if you want to try out different configurations or layouts. The editor feels like a mature development environment and is a pleasure to work with.

Changes made to a preset in the Web environment are not immediately reflected on the hardware. However, a simple push of a button and the hardware is updated in seconds, so there is no discernible gap between editing and testing in the real world.

SysEx & LUA Scripting

SysEx, any intrepid MIDI explorer will tell you, is a pain in the backside to work with. But the Electra One has a set of editors and systems designed to make it slightly less so. A visual SysEx editor is used to set up templates. You use templates because SysEx strings tend to repeat a lot. You can work in hexadecimal, binary or decimal, and the system will convert to whatever is necessary. There's also a Learn function which listens to the MIDI inputs and learns the message type, making it easy if your chosen MIDI device sends MIDI for the relevant parameter.

There are also tools to deal with preset SysEx dumps (as opposed to individual parameters). You can inspect patch SysEx dumps and map parameters to the relevant byte positions within. Again, the MIDI Learn function helps. The result is that the Electra One can receive a MIDI SysEx patch dump and update all on-screen parameters accordingly. There is one caveat to this, which is that there is no real way to deal with bulk dumps of multiple presets at once. But this is still one of Electra One's defining strengths. The ability to map SysEx dump bytes directly to parameters is something that many MIDI editors shy away from. My Roland MKS-50 will send out a SysEx dump of the new preset whenever you change presets. The Electra One will receive this and parse the SysEx, updating all the on-screen controls to the correct values for the current patch.

What On Earth Is LUA?

LUA is a popular scripting language used in gaming, industrial applications, and some audio applications you might already have heard of, like Reaper, Reason and Kontakt. It is optimised for embedded systems, meaning you can run it on hardware. Electra One's LUA implementation has an impressive API (Application Programming Interface). An API is a library of pre-written functions and callbacks you can call upon that are specific to the task at hand — in this case, MIDI control and the manipulation of the Electra One hardware. Using scripting, you can monitor the actions of the hardware, intercept controls, alter anything on the screen, manipulate MIDI messages, and much more.

The online Electra One documentation is exhaustive and impressive. It includes detailed information on the API, LUA implementation, SysEx, and MIDI handling. It's beyond my ability to describe all the possibilities of the API, but it handles every aspect of the Electra One hardware and the MIDI it can send and receive. In a few paragraphs, I'll give examples of what LUA scripting can do and leave it there because a full dive into the complexities would fill a book.

There is one LUA script per Electra One preset. This is where you write all your custom functions. Each function is named, and you can call functions from different places. For example, every control in a preset can call a function whenever it is changed. This allows you to interpret and display the data differently if you wish. You can also call functions whenever MIDI is received, whenever a new preset is loaded, when a page is changed, when a particular knob is moved, or button is pressed.

The Web MIDI development environment includes a script editor, debugger, MIDI console and JSON editor (JSON is the file format used to store data). With these tools, you have everything you need to write scripts, debug them, and monitor the MIDI inputs and outputs. A preset's LUA script is an integral part of the preset and works fully from the hardware once the preset is updated and transferred across.

In Action

During my time with the Electra One, I wanted to experience creating a preset controller from scratch. I have good »

Librarian?

I mentioned during the early paragraphs of this review that the Electra One could function as a librarian. That is perhaps only partially true. A click of the middle button on the right will bring up the Snapshots page. A Snapshot is a saved state of the current preset editor. You can save up to 432 Snapshots per preset, organised into 12 banks of 24. If you have every parameter of a synth mapped in the controller, you could use the Electra One to store and recall synth presets.

Snapshots can be saved, recalled and deleted from the hardware unit, and the Web MIDI editor also allows you to rearrange, rename and colour them. I said that it only partially qualifies as a librarian because, to fully qualify, it should be able to receive and manipulate bulk dumps of multiple presets from synths, samplers or other MIDI equipment. Currently, there is no way to achieve this. However, Snapshots go some way towards that goal, and the Electra One online manual does mention that the Snapshots editor is still in development, so more features may be added in the future.

Drag and drop to create your ideal MIDI controller, then make it look pretty with colours and groupings.

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knowledge of MIDI but have not done any programming since my BTEC in Computer Science 25 years ago. I have occasionally tinkered with scripts in other languages (Kontakt's KSP, for example), but LUA was new to me.

I started by downloading and studying a few presets from the Electra One preset library. Even though there was already a preset for my Roland MKS-50 in the Electra One library, I decided to create one from scratch, allowing me to add some missing features and lay it out differently.

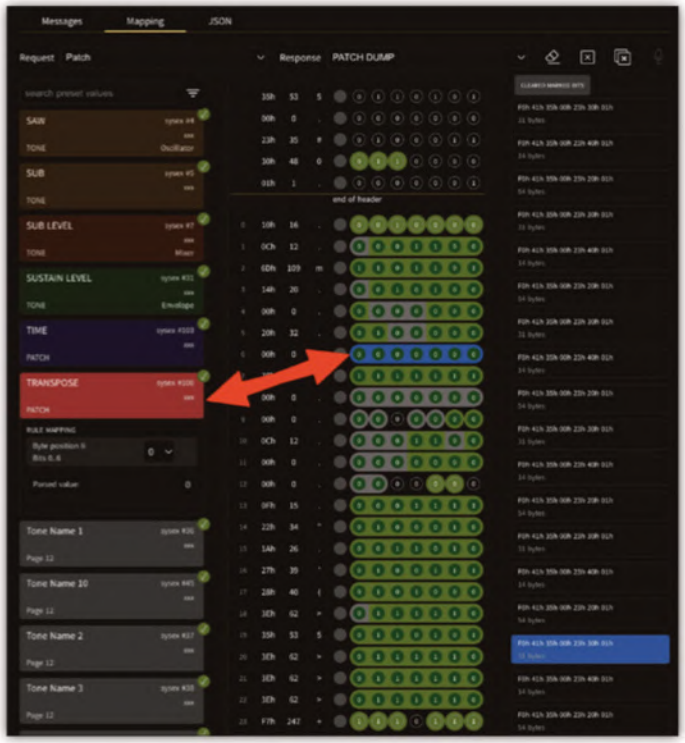
The MKS-50 uses SysEx to communicate. Thankfully, Roland's manual has a comprehensive SysEx MIDI implementation chart, so I studied that and figured out the correct SysEx strings for most parameters.

Using the Electra One SysEx editor, you set up a basic SysEx structure, which is repeated for all parameters, and then simply change any bytes to suit the parameter as needed. The process took a while to figure out. I would label it as intermediate-level difficulty.

The process of drag and drop to lay out the pages and parameters is simple, and I quickly had a configuration I was happy with. After figuring out the SysEx strings, assigning them to the correct controls was simple enough. So far, I had two pages of parameters with a combination of sliders, buttons, customised lists and even waveform graphics copied from the Roland manual.

It was time to check out some scripting. First, I wanted to alter the visibility of some controls depending on the state of others. The MKS-50 can operate in three polyphony modes: Mono, Poly or Chord Memory. In Chord Memory mode, its six voices can be stacked into a chord with each voice being transposed. I only wanted these six voice-transpose parameters to be visible when in Chord Memory mode, and hidden when in the other two modes.

As a non-programmer, my first instinct was to check out some other LUA scripts to see if anyone else was doing something similar. Sure enough, I found a similar function on another preset from the Electra One preset library. It was easy to copy and adapt the script for my use. I had a custom function called chordMode, which, when called, would receive the current value of a parameter (the polyphony mode) and set the visibility of other parameters based on that. The code for this function was seven lines long. The last step was to put the function's name (chordMode) into the inspector for the polyphony mode parameter so that it could be called whenever that parameter was changed.



Electra One's SysEx mapper helps you interpret SysEx dumps. Here, we can see where the MKS-50's Patch Transpose parameter is mapped inside a SysEx dump.

I noticed in the same preset that I'd found the hide/show script, another function for displaying MIDI notes instead of numbers. It's a simple four-line function that takes a parameter number, say 61, and converts it to a MIDI note name: C#3. I called that function for any parameter that used notes, such as the Key Range for the MKS-50's patch.

The next bit of scripting was a bit more involved. I mentioned earlier that two-way communication is a big part of Electra One's power, and I wanted to implement this in my MKS-50 editor. Although this doesn't necessarily need scripting, I wanted to display the patch name on screen, which would require a custom function.

First, I had to teach the Electra One what the MKS-50 preset SysEx dumps look like and how to interpret them. Once again, the Roland MKS-50 manual was invaluable.

The final LUA script for my MKS-50 editor.

```
-- Roland MKS-50 functions for Electra One --

deviceId = 1
local toneName = ""
local patchName = ""
local tonePage = pages.get (1)
local patchPage = pages.get (2)

-- Set Chord Memory and Chord Notes visible when Chord Mode is ON --
function chordMode(ValueObject, value)
  local hidethese= {58,59,60,61,62,63,52}
  for i= 1,7 do
    control = controls.get (hidethese[i])
    if (value==1) then control:setVisible(true) else control:setVisible(false) end
  end
end

-- Format MIDI notes instead of numbers --
function displayNotes (valueObject, value)
notes = {"C-2", "C#-2", "D-2", "D#-2", "E-2", "F-2", "F#-2", "G-2", "G#-2", "A-2", "A#-2", "B-2"}
  return (string.format(notes[math.fmod(value, 12) +1], (value// 12)-1))
end

-- Gets Tone name and Patch name from sysex dumps and displays them as Page names --
function patch.onResponse (device, responseId, sysexBlock)
  local blockLength = sysexBlock:getLength()
  local mks50Chars = {"A", "B", "C", "D", "E", "F", "G", "H", "I", "J", "K", "L", "M", "N", "O", "P", "Q", "R", "S", "T", "U", "V", "W", "X", "Y", "Z",
  "a", "b", "c", "d", "e", "f", "g", "h", "i", "j", "k", "l", "m", "n", "o", "p", "q", "r", "s", "t", "u", "v", "w", "x", "y", "z", "0", "1", "2", "3",
  "4", "5", "6", "7", "8", "9", " ", "-"}

  -- For Tone dump only
  if (responseId == 0) then
    toneName = ""
    print ("Sysex message received = " .. blockLength .. " bytes long") -- should be 54 bytes
    print ("Sysex message responseId = " .. responseId)
      for i= 0,9 do
        toneName = toneName .. mks50Chars [sysexBlock:peek (i + 44) +1]
      end
    tonePage:setName ("TONE: " .. toneName)
    print ("Tone Name changed to " .. toneName )
  end

  -- For Patch dump only
  if (responseId == 1) then
    patchName = ""
    print ("Sysex message received = " .. blockLength .. " bytes long") -- should be 31 bytes
    print ("Sysex message responseId = " .. responseId)
      for i= 0,9 do
        patchName = patchName .. mks50Chars [sysexBlock:peek (i + 21) +1]
      end
    patchPage:setName ("PATCH: " .. patchName)
    print ("Patch Name changed to " .. patchName)
  end
end
```

Every time the MKS-50 patch changes, it sends out a full SysEx dump of the new patch. I used the Electra One Learn function to capture a typical SysEx dump, and then used its mapping tool to map all the parameters I had already set up to specific bytes inside the SysEx dump. This is venturing into the advanced side of the Electra One's capabilities. It wasn't easy to learn, and it took some trial and error to get right, but I got there in the end. The result is that the Electra One always shows the correct values for all parameters when you change a preset, which is something even the original dedicated Roland PG-300 MIDI controller could not do.

The finishing touch was to script a function to be called whenever a SysEx dump was received. This function would look in a specific location within the dump and extract the patch name to display on screen. Roland didn't include any SysEx to rename patches remotely; otherwise, I would have liked to include that in my custom MIDI controller as well, but these are the limitations sometimes. Even though I can't rename patches from the Electra One, it's still very satisfying to see which patch is loaded without looking at the synth.

Overall, I ended up with a highly functional, specifically customised MIDI controller for my Roland MKS-50 that is as good as any commercially available controller for that synth, if not better. Making your own controller forces you to better understand your equipment, and the result is customised to you. I was blown away by how well the Electra One worked and adapted to this particular synth. The power of LUA scripting and the excellent Electra One API, which allows you to manipulate both MIDI and the Electra One itself in a huge variety of ways, is very impressive.

Conclusion

The Electra One has the power to be useful to almost anyone with a MIDI setup. It's still a very versatile controller, even if you never touch the LUA scripting or the SysEx editor. Some people might not take to the touchscreen, and might want more controls, but I never really felt like having 12 knobs was a barrier to quick editing. And the touchscreen is almost entirely optional if you're more of a button pusher.

Even after a few months of having the Electra One around my studio, I'm still discovering new ways to use it. For example, you can pair it with something like an Elektron Octatrack to extend the functionality and give you more knobs to tweak during a live performance. The Electra One could be an amazing live performance partner for many reasons.

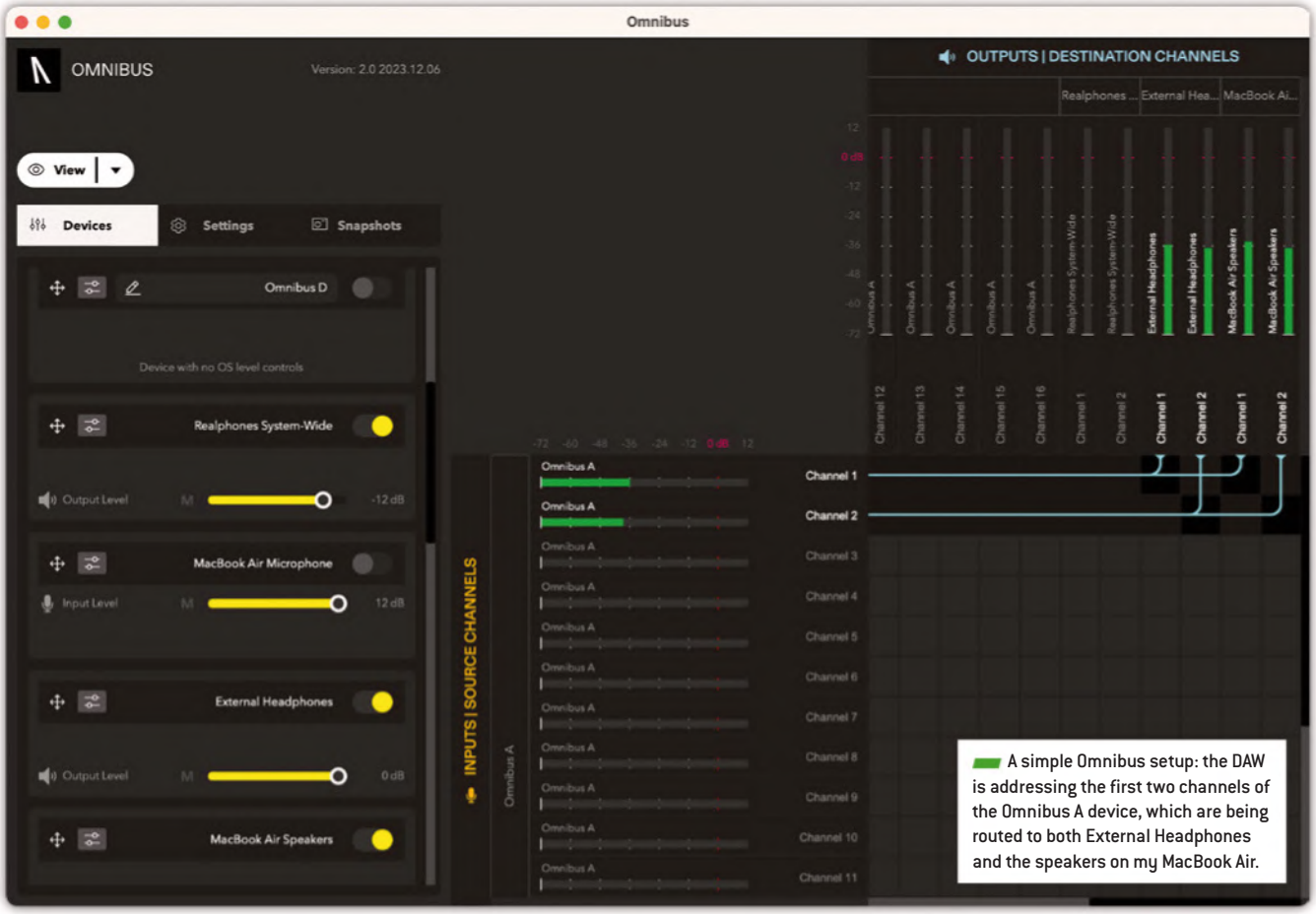
It's also useful if you want a DAW controller. There are some quite sophisticated presets in the Electra One library for Ableton, Bitwig and other DAWs that have full two-way communication, live clip launching, device awareness, and transport controls. If you don't have space on your desktop for something like the Ableton Push, the Electra One can give you some of the same functionality in a much smaller space.

The Electra One, apart from being a MIDI Swiss Army knife, is also a great way to teach yourself some coding. And those that already know their stuff will very quickly be reaping the rewards. I've been very impressed by its abilities, and I've barely scratched the surface. It will definitely be staying in my studio to help fight the good MIDI fight.

$ €553
W www.electra.one

Audiomovers Omnibus

Virtual Patchbay & Routing Matrix For macOS



A simple Omnibus setup: the DAW is addressing the first two channels of the Omnibus A device, which are being routed to both External Headphones and the speakers on my MacBook Air.

Omnibus takes audio from A to B, C, D and wherever else it's needed!

SAM INGLIS

If Audiomovers is a familiar name, it's most likely because of ListenTo, which allows audio to be streamed over the Internet in real time. Abbey Road Studios liked this product so much they bought the company, having been told by their own engineers that ListenTo was the only solution they trusted for remote session work. But ListenTo is not the only problem-solver in the Audiomovers line-up, and even if you never need to send audio anywhere outside your own studio, tools such as Omnibus and Inject could prove invaluable. The latter is a cross-platform utility that allows audio from secondary USB devices to be integrated into a DAW environment through a plug-in insert, making it possible for example to record audio from a synth's USB output whilst the DAW is still addressing your main audio interface.

Omnibus, by contrast, is Mac-only, since it engages with Core Audio in a fairly deep way. In essence, it's a routing matrix and aggregator that allows audio to be routed freely between all the different virtual and physical Core Audio devices in your system.

Driving The Bus

A small download from the Audiomovers website, Omnibus is authorised online, and gives you the choice as to whether it should run automatically when you boot your Mac, or wait until it's called upon. It creates four virtual drivers of its own: the 16-channel Omnibus A and B, the 32-channel Omnibus C and the 64-channel Omnibus D. These are addressable just like any audio device, from the Sound tab of System Preferences and from any Core Audio applications on your Mac.

Omnibus itself is a single-window application, though its single window can end up being pretty large, especially if you have other virtual drivers such as the Pro Tools Audio Bridge installed. By default, the left-hand panel shows a list of all the devices currently active on your Mac; this will include the four Omnibus drivers, system audio such as External Headphones and built-in speakers, 'systemwide' programs such as Sonarworks Reference or dSoniq Realphones, and any connected Core Audio interfaces (hardware that also uses other driver protocols, such as Pro Tools HDX, can only be accessed through its Core Audio drivers). If a device presents input and output level or mute controls that can be changed within macOS, they'll appear here. Each device also has a toggle that adds or removes it from the main patchbay.

Two other views are available for this left-hand panel. Settings allows you to define a global sample rate and buffer size for Omnibus itself, and provides various other global configuration options. Snapshots lets you define up to 100 presets, nine of which can be assigned to hotkeys for instant recall.

The main part of the Omnibus window is taken up with the patchbay, which works pretty much like most other virtual routing matrices. Source channels are displayed on the vertical axis and destinations along the horizontal axis. To make a connection, you simply click, whereupon a blue line appears showing how the two are linked. A key Omnibus feature is that this matrix supports multing and combining in practically any configuration: there doesn't seem to be a hard limit on how many sources you can route to a single destination, or *vice versa*. They don't even have to be at the same sample rate: Omnibus quietly takes care of things like sample-rate conversion behind the scenes.

Wild Ride

Some of the use cases that prompted Audiomovers to develop Omnibus probably apply only to a minority of *SOS* readers. For example, in an Atmos context you could use it to route a multi-channel DAW output to your own 7.1.4 speaker system, to a separate binaural encoder, and to ListenTo for someone to audition the mix in a different country, whilst simultaneously hosting a Zoom call for remote talkback. But you don't have to think very hard to find applications that are valuable even in more humble setups.

Right at the other end of the scale, for example, one of the frustrations of Core Audio on Apple laptops is that the External Headphones device vanishes when nothing is plugged into the headphone output. So you typically have to make a trip to your DAW's audio preferences, or even close and reload the DAW session, every time you want to swap between headphones and a MacBook's built-in speakers. Needless to say, this gets old fast, but 30 seconds' worth of clicking in Omnibus completely eliminates the problem. Select one of the Omnibus drivers as your DAW's output device, plug in your headphones, and you can route the mix simultaneously to headphones and speakers. Unplug the headphones and the speakers will carry on. Best of all, when you plug the cans in again, Omnibus will remember and automatically restore the routing.

There are innumerable other ways in which Omnibus can make itself useful. If you have a bunch of different USB audio devices connected, Omnibus can effectively aggregate them all, allowing you to track from multiple USB mics or synths at once. Want to set things up so that a systemwide headphone or monitor correction package only 'sees' one pair of outputs on your interface? No problem. Run multiple audio programs simultaneously, integrate DAWs and virtual meeting software, record audio from your web browser... the possibilities are endless. And what's most impressive about it is the sheer robustness of the environment, which handles unexpected disconnections, sample-rate changes and anything else you can throw at it without breaking sweat. If you've dabbled with aggregate and systemwide Core Audio devices in the past, you may well be nervous about using them for mission-critical roles, but Omnibus appears to be bulletproof. ■■■

ALTERNATIVES

The only partial competitor to Omnibus I know of is **Rogue Amoeba**'s **Loopback**, which is slightly more affordable but offers a rather different visual paradigm that is not based on a routing matrix.

$ $149.99
W www.audiomovers.com

Audiomovers Omnibus
$150

PROS
- Endlessly configurable audio routing between Core Audio devices.
- Supports mixing and multing.
- Very robust.

CONS
- Mac only.

SUMMARY
A supremely versatile and surprisingly user-friendly virtual routing matrix for Core Audio, Omnibus can solve problems you didn't even know you had.

Sontronics Saturn 2

Capacitor Microphone

Sontronics' unmistakeable Saturn is back — with digital pattern control to complement its vintage looks.

NEIL ROGERS

Sontronics' original Saturn was reviewed way back in *SOS* October 2010, when Paul White found much to like about this distinctive, retro-looking large-diaphragm capacitor microphone (www.soundonsound.com/reviews/sontronics-saturn). Despite it being a very successful product, Sontronics made the difficult decision to officially discontinue the mic in 2015, as a key component of the circuitry was no longer available or easily replaceable. The microphone was close to designer and Sontronics founder Trevor Coley's heart, so it was always the plan to re-release the Saturn at some point. A new version was seemingly set for release just before the pandemic struck in 2020, which put their plans on hold. Finally, nearly a decade after the original Saturn was discontinued, the Saturn 2 is now available, and I was very much looking forward to putting it through its paces in my studio.

Polar Exploration

The Saturn 2 is hand-built in the UK, and comes with a reassuring lifetime guarantee. Sontronics' aim with this new version was to preserve the character of the original Saturn, whilst seeking to enhance it wherever possible. As such, the sonic nature of the capsule remains broadly the same, with the headline change

Sontronics Saturn 2
$1999

PROS
- Highly capable all-round studio mic.
- Stunning results on the right singer.
- Plentiful, easily controlled polar pattern options.
- Well-judged high-pass filters.
- Comes with an excellent case!

CONS
- Mount system is frustrating to get in place.
- Polar pattern defaults back to cardioid when powered on/off.

SUMMARY
After a longer-than-expected hiatus, Sonotronics' popular Saturn is back in orbit. The Saturn 2 is a highly capable, high-end studio mic that features a new digital switching system for its now expanded polar pattern options.

Hear For Yourself

To hear how the Saturn 2 performed on a range of sources, check out the audio files available to download at https://sosm.ag/sontronics-saturn2-media.

being the new digitally controlled internal electronics — which, Sontronics explain, have been painstakingly engineered to provide extremely close control over the voltages sent to both sides of the capsule. These accurate voltage adjustments allow precise adjustment of the polar pattern, and whereas the original offered a choice of cardioid, subcardioid, hypercardioid, omni and figure-8, this new version includes four additional patterns that sit between the five main options. With a total of nine polar patterns to choose from, the Saturn 2 should more than satisfy engineers who like to dig deep into really fine-tuning their mic's directional response. Another key benefit of the new digitally controlled design is the consistency with which the mics perform. Different Saturn 2s are precisely matched out of the box, which is obviously important for anyone planning to use multiple mics in stereo or immersive arrays.

First Impressions

It's perhaps important to point out that I have never used the original Saturn, so my review will be very much about assessing this new release on its own merits! Like other Sontronics mics I've encountered, the Saturn 2 comes in a large, velvet-lined briefcase. This feels a little over the top for a studio mic, but I'm very much OK with that, especially since many other mics now seem to ship in nothing more than a flimsy cardboard box. The mic itself is just as visually striking as its lavish case, and I was struck by just how large the cylindrical frame that supports the suspended capsule is. The design is undoubtedly vintage-looking, but it also feels modern and utilitarian at the same time — it looks good to me, and a number of clients who came through the studio thought the same.

I'm the owner of a pair of Sontronics Sigma ribbon mics, and although they get used a great deal, you will often hear me grumbling every time I have to put them in and take them out of their mounts. The new version of the Saturn features the same mounting system, and whilst it works perfectly once in place, it takes a frustratingly long time to screw the mic into the mount in a busy session.

As well as the plethora of polar pattern options, which we'll look at in a moment, the mic features two easily accessible high-pass filter settings — 75Hz and 120Hz — and the option to pad the output signal by either 10dB or 20dB.

Smooth Cymbals

One of my favourite techniques for recording drums at the moment is to have a spaced pair of ribbon mics covering the left and right sides of the kit, combined with a capacitor mic acting as a 'mid overhead' to capture more of the upper mids of the snare and toms. This seemed like a good place to start with the Saturn 2, and I positioned the mic so that it was above the drummer's head, but safely out of the range of his sticks! I was immediately impressed with how smooth the top end of the cymbals sounded, and I also took advantage of the hypercardioid polar pattern option to focus the sound more on the snare drum. The digital switching between polar patterns felt very smooth and intuitive in use, but I should point out that the mic always defaults back to cardioid when phantom power is switched off and then on again. This particular drum session took place over two days — with all the mics left in situ overnight — and I forgot to reset my chosen polar pattern on day two! Staying briefly on the drum theme, I liked how the Saturn 2 sounded as both an overhead and mono room mic in my live room over a few different sessions; I always had a sense of a subtle high-end roll-off, which worked superbly for keeping cymbals sounding smooth and contained.

Singers & Guitars

My next session was with a folk singer who always wants to record playing her acoustic guitar and sing at the same time. I like to use two mics in figure-8 in this setting, with each mic's nulls pointing towards the opposite source, to get some isolation and control over the voice and guitar. Starting with the Saturn 2 on vocals, it sounded perfect for this singer's voice, with a lovely, open, detailed sound that literally made the hairs on my neck stand up. Suitably encouraged, I experimented with seeing how close I could get the mic on one of the quieter songs in the session, and I found myself using the 125Hz cut to filter out the enhanced proximity effect you get in figure-8. It worked superbly, capturing an intimate vocal performance.

As an experiment towards the end of the session, I asked the singer if she wouldn't mind doing a quick take with the mics reversed. On a fingerpicked guitar part, you could hear a slight presence peak on the Saturn 2 — at around 7kHz, to my ears. Comparing the Saturn to a mic I know very well indeed was helpful to me and, generally, the Saturn came across as pushing forward the upper midrange presence of a source slightly — which was either good or bad, depending on what was in front of it. Another thing I noted was that the Saturn 2 needed dramatically less gain than my vintage U87, and was extremely quiet even when I cranked the preamp as loud as I dared.

A True All-Rounder?

The Saturn 2 is billed as a high-quality studio mic that will be at home in many different settings. Due to its physical size, it isn't going to be a snare drum mic, but I observed nothing that would contradict that claim otherwise: it got heavy use in my studio on piano, drums, various acoustic guitars, vocals and even a voiceover session. As I mentioned earlier, I was often aware of the subtle presence peak, but for the most part, it felt like it presented what was in front of it in a neutral but ever so slightly flattering way — due in part to the pleasing sense of the very high-end frequencies sounding rolled off. I'm not an engineer who gets overly fussy with polar patterns, but with the switching being so easy (I often got an artist to change the pattern for me!), I found myself playing with subtle changes in patterns, and I especially liked nudging around between the omni and cardioid options when recording acoustic guitars. I imagine that over time I could get even more involved with the options, and I also found myself using the high-pass filter settings more than I do on some other mics.

Sontonics' higher-end mics have always done well, offering up a credible alternative to many of the well-known vintage mic options. You get a great-sounding, British-made mic that is quiet, has a healthy output level and won't misbehave mid-session or need regular servicing. This has certainly been my experience of the Sontronics mics I have owned, and it's true for the Saturn 2. This does face some stiff competition in this price range, but I found it to be a great-sounding, flexible addition to my studio mic collection that helped me make some memorable recordings during the review period.

$ $1999
W www.sontronics.com

With the instrument's image clipped by the sides of the window, the CP-70 V GUI is a rather plain affair that doesn't hint at the quality of the sounds that it can produce.

Arturia V Collection X

Software Instruments

With new and improved instruments Arturia's one-stop synth shop is better value than ever.

GORDON REID

The inexorable expansion of Arturia's V Collection continues unabated, most recently from 2021's V Collection 8, to 2022's V Collection 9 and, now V Collection 10... umm, I mean V Collection X, which continues the company's tradition of including some new instruments, some upgraded ones and, on this occasion, a bunch of new sounds contained within three expansion packs. The new instruments comprise CP-70 V, MiniFreak V, Acid V, Augmented Brass, Augmented Grand Piano and Augmented Woodwinds. 'Hang on,' I hear you say, 'Wasn't MiniFreak V reviewed in *Sound On Sound* in April 2023, and wasn't Acid V reviewed in November?' Of course they were, or I wouldn't have mentioned it. Augmented Brass and Augmented Grand Piano were also released last year, so only two of the instruments in V Collection X are actually new. This doesn't make the Collection any better or worse value, it's just that, when Arturia claim that six new instruments have been added, they mean new to the Collection, not new as in... well, new.

New: CP-70 V

The first of the two new instruments is perhaps the most important addition to the Collection, so let's start with CP-70 V, a recreation of the Yamaha CP-70 piano. You might wonder why Arturia didn't emulate the Holy Grail of the CP series by sticking an 88-note keyboard on the GUI and calling it CP-80 V, but that would be to overlook that the two original pianos are rather different: the CP-80 has a larger harp with longer strings so its tone is rounder and deeper, and — notwithstanding the extra notes — it feels subtly different to play.

CP-70 V isn't physically modelled; the initial sound generation is sample-based. This wouldn't have surprised me 10 or 20 years ago but, today, I suspect that might have been a decision made if the company found that they couldn't get the results they wanted from physical modelling. Nonetheless, modelling was used to emulate the original's electronics, and Arturia claim that the combination of the two approaches "allows for a truly expressive sound that perfectly recreates this remarkable instrument".

Of course, no two CP-70s are the same, so the company have added an Advanced panel to CP-70 V which offers 14 parameters that allow you to tweak the emulation to obtain the characteristics that you want. Once programmed to taste, the resulting sound — which can be mono or spread across the stereo field with bass notes to the left and treble to the right — passes through an effect section that contains four slots hosting 13 freely assignable effect types, plus two dedicated slots that contain an amp modeller and a reverb.

Creators of CP emulators tend to iron out the worst idiosyncrasies of any specific instrument in order to produce a more polite and well-behaved sound; this makes the emulation more widely useful, but can reduce its character when compared with an original instrument. So if there's an immediate difference between CP-70 V and a vintage CP-70, it's that the tuning — even when you dial in a bit of detune — is more accurate than anything other than a recently tuned original. Furthermore, the experience of playing it from your MIDI keyboard is unlikely to be the same, because the original's keybed is that of a Yamaha acoustic grand piano. Nonetheless, you'll get very close with a bit of tweaking and, when playing it from an expensive, fully weighted controller, the differences are unlikely to matter unless you're a true CP-70 fanatic.

Before moving on, there are two further points to note. The first concerns the GUI. Unlike many previous Arturia soft thingies, the image of the instrument fills the whole window, and is even clipped at the left and right edges. I've noted in the past that the company's GUIs could take up a lot of space, but I don't think that this is an attractive way to address the issue. The second concerns my single, brief attempt to emulate my CP-80 using CP-70 V. In short, I couldn't. This wasn't only because the underlying tone is different, but because Arturia have declined to extend the uppermost and lowermost samples beyond the range of the CP-70. I don't blame them for this, because I could easily believe that these notes would sound inauthentic.

New: Augmented Woodwinds

Back in 2022, I suggested that the introduction of Augmented Strings and Augmented Voices in V Collection 9 might herald the introduction of further Augmented products, and so it has proved. Augmented Woodwinds is the latest in the series and, in many ways, it's identical with the others, sharing the same eight controls on its front page, the same four-source/dual-layer sound generation, plus the same filtering, modulation, arpeggiation and effects. The difference between the various Augmented products therefore lies in nothing more than the sample libraries that underpin their main sample engines.

Despite its name and a library that focuses on solo and ensemble woodwind instruments, Augmented Woodwinds isn't an instrument that I would use to create orchestral performances in the way that I might use, say, a Kurzweil or Korg sample library. Instead, I found its floaty leads, airy pads, doom-laden carpets of sound, sparkling fairy dust and sound effects to be ideal for film and TV soundtracks and general 'ooh, I didn't expect that' sound design. As with previous Augmented instruments, I hadn't expected too much from it but was again was surprised by what I conjured quickly and easily when I dived in and, the more I used it, the more I respected it. It's unlikely to be your main soft synth, but I suspect that it will take your initial ideas in unexpected directions, and I'm sure that it will be another useful addition to your armoury.

The eight controls on the top level of Augmented Woodwinds will look familiar to users of existing Augmented instruments.

Revised: Mini V4

Over the years, Minimoog V has evolved from its first, slightly iffy incarnation to the power and flexibility of Mini V3, so it's perhaps surprising to see Arturia's designers rein it in for Mini V4. They've reduced the maximum polyphony from 32 to six voices, removed pulse-width modulation and sync from the oscillators, »

Arturia V Collection X
$599

PROS
- It features two new instruments.
- Four more from the existing Arturia family have been adopted.
- Two more have been updated, including the flagship Minimoog emulation.
- Arturia's soft synths keep improving, and they can sound superb.
- At just $13 per instrument, it's remarkable value for money.
- You don't need a warehouse in which to store everything.

CONS
- The new GUIs are not always as attractive as previous instruments.

SUMMARY

It's a huge collection of high-quality soft synths and other instruments, and ridiculous value for money if you're entering this soft synth lark for the first time. Upgrade pricing is available and, if you just want a handful of the products contained, you can still buy each one individually.

The Advanced pages of the Mini V4 show that, while simplified, it still offers a lot of sound shaping potential.

» removed the ability to disconnect osc 2 from the keyboard, removed the motion recorder and reduced the modulation capabilities, all of which reduces the range of sounds that you can obtain from it. The company have sought to justify this by saying, "We have stripped back the feature set to try and be as pure as we can to the original synth", but I doubt that this noble philosophy is the real reason for all of the changes since many additional facilities have been retained or even enhanced. I'm much more inclined to believe that the new algorithms that underpin the oscillators, mixer, filter, contours and feedback are hungrier than those in earlier incarnations, so it was necessary to slim down the feature set to make V4 run reliably on anything but high-powered and therefore expensive computers.

Given all of the omissions, is there anything new to shout about? Indeed there is and, while there are too many changes to discuss here, I'll concentrate on the most important. First, there are the two new types of feedback (which replace the External Input loop of V3) plus overdrive. Next comes a filter Emphasis button, which allows you to replicate the response of the original Minimoog (in which low frequencies are attenuated as you increase the Emphasis) or that of some other vintage synths that don't suppress the low end when the Emphasis (or 'resonance') is increased. The third major improvement is the addition of a Vintage knob and Dispersion, which allow you to determine the amount by which various aspects of the sound generation differ from voice to voice. Finally, the effect section is more flexible than before.

So, what of the underlying sound? Has it improved enough to justify all of the omissions? I think that it has. Even without invoking any Dispersion, V4 is subtly more organic than V3 and (while I dislike the overuse of these terms) it can sound creamier and fatter. Only you can decide whether the compromises represent a step forward or backward, but I don't need my Minimoog soft synth to offer massive polyphony and oodles of modulation routings; I need it to sound as good as it can. On this basis, I would use V4 in preference to any earlier version. There's just one additional caveat. The changes mean that earlier Mini V patches are incompatible with the new version. Fortunately, I found nothing to stop me running V3 and V4 alongside each other so, if you have the older version, all of the work that you put into creating your existing patch libraries isn't wasted. But if Mini V4 is your first Arturia Minimoog, I suspect that you'll want to start programming right away, especially since the number of factory patches has been reduced from 460 in V3 to 161 in V4.

Revised: Wurli V3

I must admit that I became a bit of a fan when I reviewed Wurli V2 because it barks beautifully in that rather curtailed way that Rhodes pianos just don't. In addition to its eight underlying models, its advanced controls gave me access to things such as the vibrato circuit, pickups, hammers, dynamics and EQ, making it possible to obtain all manner of variations from what, on the surface, looked like a simple replay instrument. Adding auto-wah at the start of the effects chain and one of the amplifier/microphone models at the end placed the Wurli world at my fingertips.

Behind its revised GUI, the new version embodies many changes. Most noticeably, the Advanced panel offers more parameters to fine-tune the character of the instrument, and the effects section is again expanded. Arturia also claim that the sound engine that underpins V3 has been updated from its predecessor. But is the new version better? To be honest, it's hard to tell. The factory sounds (of which there are now just 51 rather than 190) are different and, when I tried to recreate some of my V2 sounds on V3, I failed because the two don't map from one to the other. For example, while the effects section in V3 is more flexible and its new GUI makes the controls and labels on each stompbox clearer, its three-band parametric EQ doesn't allow me to shape the sound in quite the same way as the graphic equaliser in V2. Ultimately, testing Wurli V2 and V3 next to one another was much like playing two EP200s — they're similar, but individual. Both versions can sound great and can respond as you would want so, if you have V2, I recommend that you hang on to it when you install V3, and then choose whichever works best for you on any given track or recording.

Conclusions

As in previous reviews of V Collections, this has by necessity been a whistle-stop tour of the major changes. So let's finish by considering the collection as a whole. Now containing a staggering total of 39 instruments that cover vintage monosynths, revered polysynths, samplers (including the Emulator II, Fairlight and Synclavier), Mellotrons, pianos, Clavinets, organs and the company's own sample-based synths,

The Wurli V3 FX page eschews the 'stompboxes on the floor' imagery of the previous version and offers a much clearer (if less evocative) representation.

it's likely that you'll be able to obtain almost anything that you want from it. Well... almost anything, other than state-of-the-art acoustic and electronic drum kits. At a price of around $500, that equates to around $13 per instrument, which is ridiculously cheap. So, given the high prices and low reliabilities of many of the vintage instruments that the Collection seeks to emulate, the only questions for many players will be, do the instruments sound good, are they easy to use and — where appropriate — are they accurate? The answers are yes, yes and, despite the protestations of the Analogue Anoraks (who haven't yet learned that decrying soft synths doesn't make them sound cleverer than the rest of us), yes. It's not all-encompassing, and I'm still hoping for a couple of additions that I requested from Arturia a few years ago, but there's a valid argument to suggest that, if you can't get what you want from V Collection X, you must be on esoteric sonic ground indeed. Either that, or you simply haven't dived in and tried hard enough! ■■■

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Midicake Arp

MIDI Arpeggiator

Midicake have reinvented the arpeggiator.

PAUL NAGLE

As the first product of Bristol company Midicake, Arp is somewhat specialised — you might even say idiosyncratic. It's billed as a four-way arpeggio generator designed to control multiple MIDI layers and produce patterns that are neither boring nor random. Without further ado, let's arpeggiate!

(Cake)Walk Through

Housed in a compact aluminium box (available in either black or white), Arp is covered in squishy rubber buttons and has a small but serviceable colour display. Its patterns are generated from notes made available from either the eight chord buttons or from an external MIDI controller, with a single octave keyboard provided for quick transposition and for programming the chords, which are initially set to major, minor, etc.

Although I'll stick with the term 'chord' throughout the review, it's helpful if you don't think of regular chords but, rather, of a pool of notes Arp can draw from. This pool can be any combination of the 12 notes of Western music and encompasses scales, modes or any grouping of your own — all relative to a root note. Incidentally, a long list of chords and scales are provided, with scales written in upper case so you can easily differentiate between a C min triad and a C MINOR scale.

There are 16 banks of 12 patches in which to store your work and all arrive populated with useful and instructive examples. Once you set each generator's MIDI channel you can begin working your way though the many parameters, copying patches to other locations as you go to create variations. Saving is done automatically, so operation is both fast and fluid.

Arp weighs 1.2kg and features 5-pin MIDI I/O, plus a 5V Type-B USB connection handling MIDI and supplying power. USB and MIDI cables are provided but, when working computer-free, you'll need to source a suitable USB adaptor plug. A USB Type-A host is also present and suitable for controllers such as the Arturia KeyStep, which it can also power.

Midicake Arp
£386

PROS
- A very different take on arpeggiation — times four.
- Easily assembled custom scales, chords and chord chains.
- Neat and self-contained unit that's even more powerful with external MIDI control and dynamic note input.

CONS
- Less immediately intuitive than a traditional arpeggiator — and there's no random direction.
- Some functionality (eg. chord chains) deserves fleshing out.
- More MIDI ports would have been welcome.

SUMMARY

A unique approach to generating layered arpeggios, chords and drones. Covering everything from easy chord progressions to Philip Glass on speed, Arp's intricate patterns can take you places you'd never reach otherwise.

As the company name suggests, Arp talks MIDI only. It's therefore a willing partner for hardware synths, VSTs or a combination of both, although the 5-pin and USB MIDI ports are not separately addressable. Unusually, each of the four generators may be assigned to any MIDI channel and configured for Arp, Chord, Drone or Pad modes on a per-patch basis.

Pressing the Set button turns the note and chord buttons from green to blue,

indicating you've entered the menu system. Menu pages are labelled on the panel and where a page has several entries, the display shows the current location. Getting around involves repeatedly pressing the button or combining it with the Up or Down buttons to go back and forwards. Parameter adjustments are made via the tempo encoder, with single increments more easily achieved with the Up/Down buttons.

Four assignable encoders that operate globally provide a direct connection to your favourite parameters without a need to enter the menu system. Call me greedy but it would have been fabulous if these had been assignable on a per-bank basis — and even better if they could have sent MIDI CCs.

When in Play mode, the display reports the active chord, bank and patch number as well as the octave selected for the button keyboard. However, much of the display is hogged by the tempo, which I didn't find terribly useful, particularly when externally sync'ed. Arp's insistence on flashing up the bpm became quite intrusive when tweaking the four macros and left me with weird flashbacks to my tempo-obsessed Roland SP-404 MkII.

The four generators are driven by a single start or stop action but each may be individually muted should you wish to thin out some of the mayhem. In addition, Follow buttons instruct the generators to play after the previous one, so if all Follows are active, the four will run in series. Right now, this process permits just one bar of each generator to play before it moves on, which, as we'll see, is rarely enough to allow an arpeggio to show its full glory.

Arpin' On

Given Arp's single MIDI output, you'll appreciate synths featuring hardware MIDI thrus. For others it's time to reach for a thru box. When you load a new patch during playback, it kicks in at the end of the bar, although if you send MIDI Program Changes from a sequencer or master keyboard, the patch loads immediately. There's no way to send Program Changes from Arp to your connected synths so you'll have to keep manual notes of any favourite combinations of Arp and synth patches — or leave it to your main sequencer or DAW.

We'll quickly run through some of the parameters available to get an idea why Arp's patterns are often unlike those produced by a typical arpeggiator. The patterns consist exclusively of MIDI notes, therefore the best results are obtained using instruments that respond to velocity.

Buttons labelled 1-4 select each generator for editing and a double-click allows relative changes to be made to all four simultaneously. There are two primary modes of operation — Arp and Chord, the latter offering two subsidiary modes (Pad and Drone). As you'd expect, arpeggios are notes of a chord played sequentially, while chords are multiple notes played at the same time. No restrictions are imposed here — a patch can consist of four chords, four arpeggios or any combination that takes your fancy.

Working through note menu buttons, you first pick a base velocity and gate length, the latter maxing out at 32 bars — ideal when programming very slow patterns. Next, you encounter the various ways of confining and limiting notes, such as locking them to a particular octave or ensuring they never stray too far from the root of the chord. You can force a generator to pick notes from a specific scale or even lock it to a single note, ignoring the input completely. This is handy for percussion or sample triggering.

Note offsets are a simple but effective way of finding parallel phrases within the current chord but for more avant garde experiments, you can bypass the chord completely by engaging Chromatic mode. This is actually a great way to quickly hear what some of the more obscure note-related parameters are doing.

Time Division is an important menu because it determines the arpeggio speed — from 32 bars per note to 32 notes per bar, with triplets thrown in at either end of the scale! Slow progressions can be a real bonus, say for tossing an occasional extra note into a shorter, more regular pattern. It was while in this menu that I noticed all parameter changes become active immediately when scrolling — there's no confirmation or 'enter' involved. That should certainly keep you on your toes during a live performance!

If Arp were a traditional arpeggiator, the number of notes held down would determine the length of the phrase. Here the Steps parameter does the same, its range from 1-16 notes. The value translates to how many steps through a chord each arpeggio should travel, so supposing the direction is upwards, a Cm triad (C, Eb and G) with a Steps value of 1 will play only C. A value of 2 will play C and Eb, 3 C, Eb and G, and so on.

In terms of playback direction, the expected Up, Down, Up&Down and so on are supported but you won't find random because Midicake are philosophically opposed to the concept. Arp's name comes from 'Automate, Repeat, Play' and a firm belief that complexity should always be repeatable. Providing you're not like me and constantly tweaking every parameter, Arp will therefore always produce the same output from a given input.

Rather less philosophically challenging, the Delay parameter is a timing offset, much appreciated when several arpeggios are layered. If you have two generators set to the same MIDI channel, the action of delaying one and giving it a lower velocity can lead to all manner of juicy MIDI delay effects.

Another key pair of parameters, Bounce and Jump, control the order that notes are selected from the pool and the intervals between them. Starting with nine default or 20 custom Bounce patterns, you'll be bouncing and jumping around producing sequencer-like phrases in no time. Lastly, Repeat is the number of times each note plays before moving on.

»

Round the back, things are pretty straightforward with USB A and B ports, MIDI I/O sockets and a dedicated button for software updates.

The next set of menus act upon the phrase as a whole. First up is Rhythm — a rhythmic template applicable to any arpeggio. With a maximum length of 16 steps, it's a vital tool for adding rests, ties and accents and generally breaking up all that relentless upping and downing. Sixteen editable rhythm patterns are available globally and, I found, are very quickly used up. In a similar vein, Groove's 16 patterns (also editable) create offsets in time; they serve up everything from conventional swing to more irregular, stumbling feels.

Moving swiftly on, there are three ways to modulate other parameters and bring your arpeggios to life. The first of these, FX, has 127 pseudo-random seeds and is easily the most 'out there' modulation source provided. FX can be applied to introduce sudden shifts in parameters such as Time Division, Step Offset, Delay, etc, and while you can't edit the patterns, tinkering with modulation amount and the reset frequency can provide some satisfyingly wacky and irregular changes.

In contrast, Mod A and B are simple LFOs with regular waveshapes and frequencies defined in (pre-selected) multiples of bars up to 32. I'd have preferred them clocked in fractions of bars or able to run unsynchronised but, even restricted as they are, their contribution is significant. The LFOs cover a much wider range of destinations than are available to FX. Indeed, pretty much every parameter is up for grabs, including MIDI channel, FX amount and Rhythm and Groove lengths and direction. Once I tried slowly sweeping the MIDI channel, causing the notes to cascade through all 16 parts of a Proteus 2000 module, I was totally hooked. It's an arpeggiator, Jim, but not as we know it!

Chained Melodies

One of the sweetest Arp features is actually far from obvious, but if you press the main encoder, you enter the realm of Chord Chaining. The chord buttons change from plain green to many different colours so you can easily identify them within the chain, which is (you probably guessed) up to 32 bars long. The chain plays a single bar of each chord in a loop until you touch a key, at which point all entries are wiped! Chord chains are a superb sketch pad but, having lost a few accidentally, I began to yearn for a means of saving them, ideally on a per-bank basis and with my eight stored chords included too. It seems like Arp is inches away from being a 16-song backing machine so I was delighted to hear Midicake are considering some means of saving chains in a future firmware update.

MIDI Control

If the onboard modulation isn't wild enough to cover all eventualities, you'll be glad of the MIDI Learn function, which assigns Arp parameters to external MIDI control. I quickly drafted in some MIDI CC output from my Elektron Octatrack in order to lure Arp down the dark alleys of probability. I went on to remotely control Arp's MIDI channel and automate precise instrument shifts for specific regions of a phrase, something that's not so straightforward on Arp itself.

External control also means you aren't limited to eight chord buttons but can dynamically populate the note pool from a connected keyboard or sequencer. Arp really shines when taking its input this way and lends itself to all manner of accompaniment duties, such as taking chords from existing songs to see what will emerge.

The Shape Of Things To Come

Prior to completing this review I had a brief chance to play with the next firmware release, and a whole new level of rhythmic complexity. Additional parameters offer much finer control over the way bars are divided up, putting polyrhythms that would baffle other arpeggiators (and most sequencers) within your grasp. If you've ever wondered what seven notes spread evenly over 23 bars would sound like, this feature is for you!

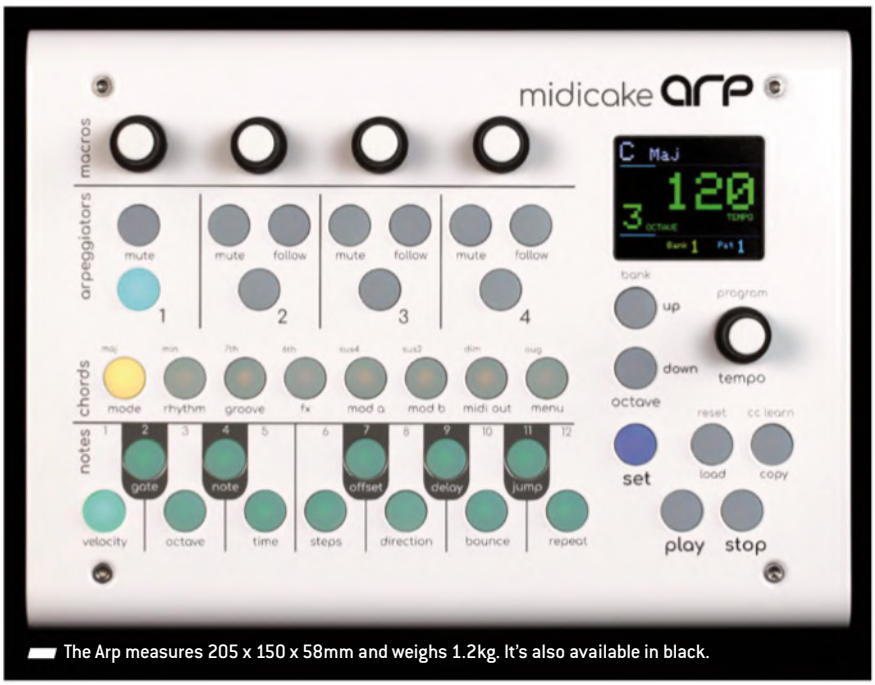
Also in this firmware is a basic means of sending MIDI CCs from the macro encoders, plus a traditional arpeggiator mode in which the length of each phrase is taken directly from the number of notes supplied.

Conclusion

It took me a little while to grasp the paradigm shift from familiar synth arpeggiators but the effort was worth it. After a few days, I realised that what is sacrificed in speed and simplicity is compensated for by the richness of the arpeggios on offer. Once familiar with the menu system, it isn't too much of a chore to navigate either.

In general, I was reminded of what can be achieved by concentrating only on note pitches, velocities and lengths, and how rewarding it can be to divide arpeggios into separate components, perhaps spread across several instruments. Equally enjoyable, you can aim all four generators at a single synth and mute and unmute parts as you play. In this way, fed either by custom chords or manual input, Arp is a flexible and interactive partner that in some cases could be an alternative to a sequencer.

Admittedly, some of its features would benefit from further development but, equally, some are so novel I can't think of equivalents elsewhere. The implementation of polyrhythms arrived almost as this review was complete but even in the short time I had, it was obvious this is going to open up whole new vistas of timing complexity.



The Arp measures 205 x 150 x 58mm and weighs 1.2kg. It's also available in black.

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Solid State Logic PureDrive Octo

Mic Preamp, A-D Converter & USB Interface

SSL's PureDrive preamps bring the sound of their Origin console to a new audience.

SAM INGLIS

When the world turned away from giant mixing consoles in the '90s and '00s, Solid State Logic were quick to adapt. Over the last 20 years or so, they've introduced lots of novel products targeted at home and project studios, while high-end mixers like the Duality and AWS Delta have incorporated advanced digital features to keep them relevant in the DAW age. So when SSL announced the Origin in 2019, it looked at first glance like an anachronism. Why would anyone want to buy a new, all-analogue, large-format console with no automation or DAW control, when you could pick up vintage equivalents for pennies on the pound?

Quite a few people, it turns out — and a second glance at the Origin makes clear that it's not at all equivalent to the analogue mixers of old. It incorporates some of SSL's greatest hits, such as the parametric EQ from their E Series consoles, and the famous bus compressor, but there's also plenty of innovation on display, including a clever new approach to routing and bussing, and a modular centre section that allows rackmounting processors, fader surfaces or screens to be built directly into the mixer. And, perhaps acknowledging the fact that vintage SSL preamp designs were never revered in the same way as their dynamics or EQ, the Origin features an all-new mic preamp circuit dubbed PureDrive.

From The Origin

The Origin has proved a deserved success, but not everyone has the space, budget or need for a large-format console. Consequently, the PureDrive preamp design has now been made available in two 19-inch rack units: the four-channel Quad and eight-channel Octo. And, in typical SSL fashion, both units add on a ton of additional functionality. The Octo was supplied for review; the Quad is described briefly in the boxout.

The core function of the PureDrive units is to preamplify signals from

Solid State Logic PureDrive Octo

$2500

PROS

- A superb clean preamp with very low noise and an epic 90dB gain range.
- Drive and impedance options introduce a useful range of different sonic colours.
- Insert points on every input.
- Stepped gain controls for precise recall and stereo matching.
- AES3, ADAT and USB digital outputs as standard.
- Excellent value for money.

CONS

- USB interfacing has limited real-world uses.
- The status of some settings isn't always obvious from the front panel.
- Settings mode can't be accessed using front-panel standby button but requires a full power cycle.

SUMMARY

The PureDrive gives you everything you'd want from a multi-channel preamp at an extremely competitive price.

mics, DI'ed instruments and line-level sources. But both units also offer built-in A-D conversion, allowing their outputs to be disgorged in ADAT Lightpipe and/or AES3 digital formats. And if doing so frees up the analogue outputs, you can then use these as insert sends, with a dedicated set of insert returns available in addition to the line inputs. The icing on the cake is the addition of a USB Type-C socket, which allows the PureDrive preamps to be used as class-compliant audio interfaces.

There are also plenty of juicy plums within the cake itself. By my calculations, the preamps offer a colossal 90dB gain range on the mic inputs, and as well as the expected high-pass filters, polarity and phantom power switches, there are also input impedance and Drive options, which we'll come to presently.

Colour Ways

Unsurprisingly, SSL have chosen not to cram all these features in a 1U chassis, but they've managed to present everything very clearly and neatly in a 2U front panel. The flip side of this is that on the Octo, most of the controls have multiple functions, with coloured LEDs used to provide visual feedback on settings. Each channel has a stepped gain control, spanning +5 to +65 dB in 6dB steps for mic signals or 0 to 30 dB in 3dB steps for line-level inputs. The first four channels also have high-impedance inputs, which are automatically selected when something is plugged into the relevant front-panel jack sockets, and offer 11 to 41 dB gain in 3dB steps. The trim control then adds a further -15 to +15 dB gain, in 1dB steps. Stepped potentiometers don't always offer the precision of rotary switches but, remarkably, SSL claim that gain values across channels are matched to less than 0.08dB at all settings. For reference, this is an order of magnitude tighter than most mic manufacturers' tolerances for pair matching.

Both the gain and trim controls also have multiple push actions. A short push on the gain knob engages the polarity reversal, whilst a long press switches phantom power on or off. The same actions on the trim knob engage the high-pass filter and swap between mic and line inputs, respectively. In the default arrangement, it's quite easy to accidentally flip polarity or introduce the filter when changing gain settings, but thoughtfully, SSL have added an optional Safe mode in which pushes of less than about a second aren't registered as inputs. I quickly gravitated to having this active all the time.

Each channel on the Octo also has two square push buttons with integrated multi-colour LEDs, and again, both have secondary functions. The ZΩ button cycles through four input impedance settings: 12kΩ (green), 1.2kΩ (not illuminated), 600Ω (amber) and 400Ω (red). These are available only on the XLR inputs, so it's not possible to vary the 1MΩ impedance that a DI'ed instrument sees, which is perhaps a shame. The Drive button toggles the selected Drive mode on and off, with a longer press switching between Classic Drive (amber) and Asymmetric Drive (green). The former introduces predominantly odd-harmonic distortion, in the same way as the Drive feature on the Origin console, while the latter is a new mode that brings in more even-harmonic saturation. Unlike the variable impedance, Drive can be applied to all input types.

The secondary function of the ZΩ button is to switch the routing so that the A-D converter for a given channel is fed from the insert return rather than the preamp output. However, this isn't activated by a long press; instead, a global Insert button lurks to the right of the front panel. Holding this causes all the ZΩ buttons to flash, whereupon you can switch the insert return for each input in or out. A down side of this approach is that although the Insert button lights up to indicate that at least one insert return is active, you can't see at a glance which channels have inserts enabled unless you press it to check — the PureDrive Quad, by contrast, has dedicated LEDs on each channel.

Also on the right-hand side of the front panel are buttons for selecting clock source and sample rate, a slightly oddly shaped display housing six-segment LED meters for each input, and a soft 'standby' power button. This is a nice touch, since the main on/off switch is on the back of the unit and thus inaccessible when rackmounted.

»

As you'd expect given that it's so well featured, the rear panel of the PureDrive Octo is pretty busy. The eight XLR mic inputs are arranged in two rows of four, while the line inputs, insert returns and analogue outputs are all on DB25 connectors. Four more XLRs carry the AES3 outputs in pairs, and there are two optical outputs, allowing full eight-channel ADAT operation at sample rates up to 96kHz thanks to the wonders of sample multiplexing. Higher sample rates are also supported, but the ADAT channel count drops to four at 176.4 and 192 kHz. Two BNC connectors offer word clock in and out, with a switchable 75Ω termination button. Mains power arrives on a standard IEC connector and the PureDrive automatically adapts to the local voltage and frequency.

See You Later, Aggregator

Connect a USB-C cable to your macOS or Windows machine, and the PureDrive Octo will present itself as a class-compliant audio interface; on Windows, you'll need to install an ASIO driver to use it with most DAW software. As it contains no D-A conversion and has no monitoring capabilities, it defaults to showing up as an eight-in, zero-out device. However, it's possible to repurpose either or both of the AES3 and ADAT outs to carry DAW outputs rather than the signals from the PureDrive's own A-D converters. At base sample rates, this could give you a total of 24 digital outputs from your DAW, which might be handy for example if you like to work with a digital mixer in the studio. For most purposes, though, it means that the PureDrive's USB interfacing is mainly useful as part of an aggregate device in Core Audio, or in conjunction with something like Audiomovers' Inject or Omnibus. It's likely to be less useful on Windows, since ASIO does not readily support aggregate devices or the use of different input and output devices simultaneously.

SSL describe the PureDrive Octo as a 32-bit interface, but it does not offer the same 'unclippable' floating-point conversion you find on some Sound Devices recorders, for example, so in practice there's no advantage to recording at 32-bit.

Like Safe mode and various other global configuration settings, digital routing is switched within a special Settings mode. This is accessed by holding down the CLK (clock) button during power-up, and requires an actual power cycle, not just the use of the front-panel standby button, so could be harder to enter if the unit is rackmounted. It's also worth noting that switching either the AES3 or the ADAT out to USB mode causes all 24 outputs to appear to your DAW, even if the other one is left in the default setting; this is so that the I/O numbering and labelling remains consistent in the DAW. Other settings options include button and meter brightness, an optional auto-sleep mode that can put the unit into standby if no control or signal input is made for a specified length of time, and even the choice of whether relays should click audibly or not when buttons are pressed.

"It's a little surprising to find that the PureDrive preamps face no obvious competition at their price levels."

Pure & Simple

If the specs are anything to go by, the PureDrive design should be a world-beater. I've already mentioned the huge gain range and outstanding gain matching between channels, and the rest of the published measurements are equally impressive. Equivalent input noise is -130dBu (A-weighted), THD+noise is below 0.0025% with Drive disabled, and crosstalk is below -108dB at 50Hz, -105dB at 1kHz and -81dB at 10kHz. Maximum signal levels that can be accepted are +21.5dBu for the mic input and +26.5dBu for the line input. The A-D converter, meanwhile, boasts a dynamic range of 119dB.

In short, if you keep the input impedance in one of the higher settings and don't enable Drive, the PureDrive very much lives up to the first part of its name. It's an exceptionally clean, quiet preamp that is enormously versatile. The high-pass filter introduces an 18dB per octave bass cut turning over at 75Hz, which seems a good choice for attenuating unwanted rumbles and thumps without affecting the main body of the signal. And, unlike the high-pass filters on many preamp-cum-audio-interface designs, it operates in the analogue domain, meaning that those noises won't rob you of converter headroom. (The filter is an area where the four-channel Quad actually improves on the Octo, as explained in the box.)

The manual states that the variable impedance settings shouldn't have any effect with capacitor mics, but I found that wasn't quite the case, at least with vintage transformer-balanced models. With a Neumann KM84, for example, the input signal was about 3dB quieter at the 400Ω setting compared with the 12kΩ setting, and very slightly fuller in the midrange. I'm not sure I could claim the tonal difference was audible, but it was just about measurable with pink noise as a source. With dynamic and ribbon mics the impedance setting makes more of a difference; using a beyerdynamic M160, I measured a 4dB overall level difference between the 12kΩ and 400Ω settings, with the latter also introducing a 2-3 dB dip at around 150Hz.

The Drive circuits, meanwhile, are commendably gradual in their onset. You don't suddenly go from clean-clean-clean to fuzzbox, as can be the case with some preamp designs; rather, there's a gentle

DB25 connectors are used to carry the eight line ins and outs, along with eight insert returns that can be substituted for the line ins on a per-channel basis.

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The PureDrive Quad

Both PureDrive units occupy 2U chassis, which means that the four-channel Quad version has twice as much panel space per channel as the Octo. SSL have taken full advantage of this, giving some things that are secondary functions on the Octo physical controls on the Quad. Phantom power and mic/line switching have their own dedicated buttons, and it also boasts a variable high-pass filter that can be swept anywhere up to 300Hz. Insert activation is now a secondary function of the gain knob, and each channel has its own LED to show whether the insert return is selected (hurrah). On the rear panel, the main outs are on XLRs rather than a D-sub, while the insert returns and line ins share a single DB25.

The four-channel PureDrive Quad has a fully variable high-pass filter and dedicated buttons for phantom power and mic/line switching.

» coloration that creeps in as you crank the gain knob. If you want to maximise the saturation, the trick is to use more gain and then back off the trim, but even so, it only gets super-hairy if you really want it to. And, as ever, it's quite source-dependent, and likely to be most noticeable on something like a DI'ed bass. The new Asymmetric Drive mode is more assertive and thicker-sounding than the Classic Drive setting. Having initially been sceptical about the idea of artificially introducing the sort of artefacts usually associated with transformer or valve-based designs, I'm definitely a convert.

If there's one PureDrive feature I didn't find much use for during the review period, it's the USB interfacing. I'd want to be very confident of the robustness of any aggregate setup before using it on a real-world session, and as long as one has a spare ADAT input, it's simpler and easier to take that approach. But there are certainly times when one doesn't have a spare ADAT input, and I can imagine it being a valuable option at a pinch.

Eight Way Fight

It's a little surprising to find that the PureDrive preamps face no obvious competition at their price levels. For half the price or less, you can pick up something like Audient's ASP880 or the Focusrite Clarett+ OctoPre; both are very capable units, but there's an understandable gap in terms of specifications and capabilities, and neither comes close to offering the same massive gain range, for example. If you're looking for something with a similar feature set, the obvious choices are Focusrite's ISA428 and ISA828, and in many ways the similarities are uncanny. The ISA preamps likewise offer a huge gain range, impressive specs, variable impedance, switched or stepped gain controls, and so on. Even the panel layout is rather similar, with the ISA828 having separate gain and trim knobs, and a bank of four instrument inputs down the left-hand side, while the analogue routing and connectivity is almost identical in both cases. In sonic terms, the main difference is perhaps that the ISA units have transformer-balanced inputs. And from the ergonomic perspective, the ISA offers a dedicated knob or button for every parameter, with no secondary functions.

The PureDrive is at heart a more contemporary take on the ISA concept, and if they were the same price, I'd struggle to choose between them. In the end, I suspect that the ISA's more immediate control set, line inputs on quarter-inch jacks and optional Dante connectivity would just about tip the balance against the Octo's Drive options and USB interfacing. But, here's the rub, they are not the same price. The ISA828 is a fair bit more expensive than the PureDrive Octo, and its digital board is a pricey cost option on top of that. I don't think the ISA828 is overpriced at all: mine has done sterling service for 15 years or more, and I'd recommend it to anyone in a heartbeat. Rather, the PureDrive Octo is a total bargain — and unless I get a sudden injection of willpower from somewhere, the review unit will soon be teaming up with the ISA in my setup. It's just the latest in a string of surprisingly affordable, over-achieving analogue processors from SSL, and I hope they're making enough margin on it to give their designers a pay rise!

$ PureDrive Quad $1399.99; PureDrive Octo $2499.99.
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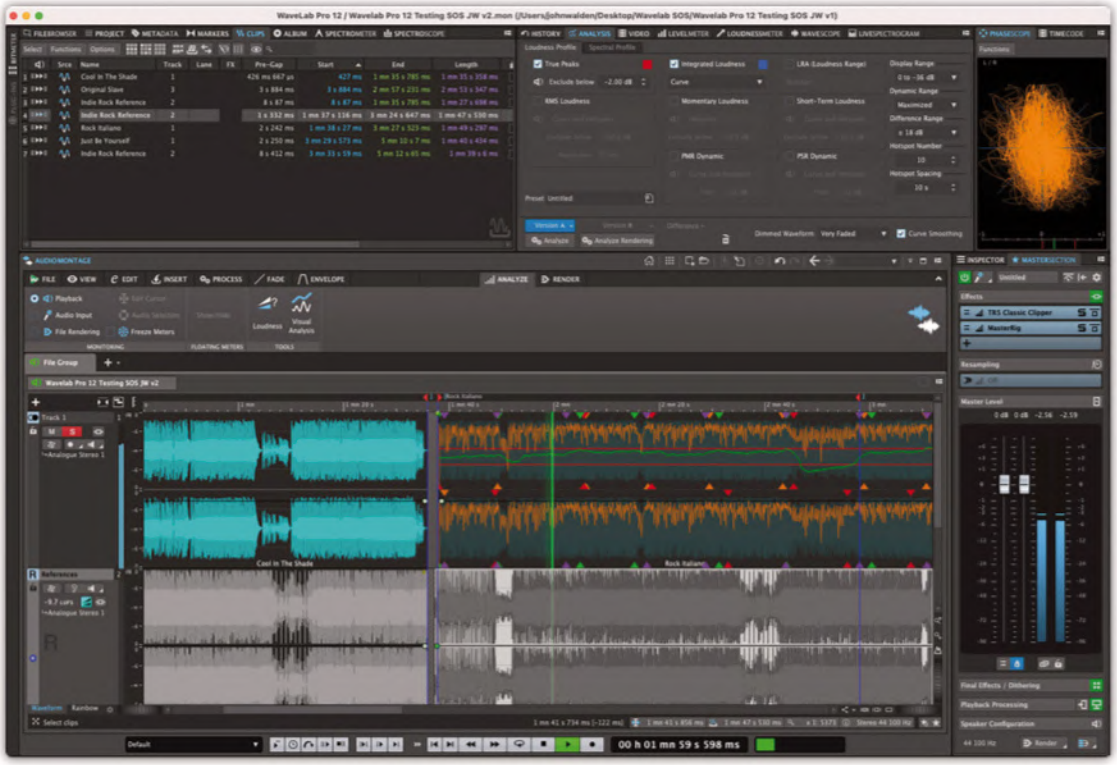
Steinberg introduce a huge number of enhancements and refinements to their flagship audio editing and mastering platform.

JOHN WALDEN



Steinberg's WaveLab has been with us for nearly 30 years. For the initial release, designer and developer Philippe Goutier's intention was to provide a powerful audio editor platform for those working with the new (at the time) sampler technology. However, over subsequent releases, WaveLab's feature list has adapted to support a wide range of additional application areas and the software now provides a very sophisticated cross-platform (macOS and Windows) audio editing and mastering suite.

Of course, many current DAWs now include their own audio editing toolsets, and there are plenty of dedicated options for audio mastering, some of which are highly automated. Outside of WaveLab's established professional user base, therefore, might some potential new users need convincing that they actually need a product like WaveLab Pro 12? Well, Steinberg are obviously keen to try, as this release offers an extensive list of new additions and refinements to what was already a substantial, feature-rich application.

So, what's new and tempting? Well, I can't fill the whole issue of *SOS*, so I'll focus here on my personal picks from the Pro 12 headlines. Time to enter the 'Lab...

Assisted Management

WaveLab Pro 12 brings a whole host of workflow refinements. I'll touch on some more of these later but two are worth mentioning from the off: the new Startup Assistant and some high-level project management options.

The Startup Assistant concept will be familiar to Cubase users. Automatically popping open when you launch WaveLab, it provides instant and easy access to any recently used Projects, Audio Montages, Audio Editor files or Batch Processing tasks, as well as your templates, audio connection settings, software documentation and online resources. Personally, I find this a very useful time-saver, but it can be deactivated if you prefer, or opened manually from the File menu at any time.

Enabled by some changes to the underlying project file structures, this release also brings some very welcome project management enhancements. The obvious highlight is the ability to create a backup of an entire Project, or a specific Audio Montage, to a new, consolidated, folder or to a ZIP archive. This is immensely useful as it ensures that copies of all the individual media files used with the Project or Montage are gathered together in a single location. Whether for data security, or to move a WaveLab session to another computer system, this brings both peace of mind and a streamlined workflow.

Differentiation & Integration

While there are clearly differences between the primary functions of WaveLab and your typical DAW, close integration between these two platforms is undoubtedly desirable. WaveLab 12 brings two significant refinements on this front. First, in both the Pro and Elements versions, there are new options to drag and drop or copy/paste audio selections from WaveLab to other applications. What's more, this can be done with, or without, effects applied to the selected audio within WaveLab.

Second, WaveLab Pro 12 can now be used via ARA within a suitable host, giving you access to WaveLab's

sophisticated audio editing toolset directly within your DAW. Support for Cubase 13, Nuendo 13, Reaper 7 and Studio One 6.5 are already approved and, while it ought to be expected, my experience in testing this within Cubase Pro 13 was very smooth.

It Might Get Loud

While most of the audio world is thankfully past the worst excesses of the loudness wars, creating appropriate loudness properties within an audio project can still be a complex task. WaveLab Pro 12 brings a number of significant new and refined options on this front.

First, new numerical and visual loudness analysis tools are available for both audio files and Montages under the Analyze tab. As shown for an Audio Montage example, the Loudness button opens a dialogue letting you generate a numerical analysis of either selected clips or the whole Montage, as well as audio held within a Reference Track. The user can customise the contents of the report and decide whether to view the absolute values or the differences compared to one of several standard targets (presets include Spotify, Netflix, EBU R128 and YouTube, amongst others, and you can also create your own custom targets). If you need to generate loudness reports (for example, to demonstrate compliance with a specific broadcast standard), the Export button then lets you do just that.

The Visual Analysis button provides similar loudness data in a visual format by superimposing curves for your selected loudness parameters over the clip's waveform display. Note that, located bottom left, this dialogue includes both Analyze vs Analyze Rendering options, allowing you to see two sets of curves, one for the original clip and one representing the clip's loudness as it would be rendered with any currently active processing applied. A 'difference' plot is also available. There is all sorts of interesting and genuinely useful functionality here. For example, you can opt to show loudness 'hotspots' (represented by triangle markers of various colours along the waveform to indicate their location) and a number of these then offer a small speaker icon. A single click then allows you to audition (and carefully listen to) just the regions around these hotspots.

Second, automatic loudness matching between your processed clips and a specified reference source is now available on playback, with either your clip or the reference able to serve as the loudness 'master'. This can serve a number of purposes, but the obvious use cases would be to compare your track in its mastered form with the unmastered version, or to compare your track with a favourite commercial release in the same genre. The automatic loudness adjustment means that, on playback, you can better judge just what your processing is doing to the dynamics or tonality of the sound while avoiding any confusion caused by 'louder is better'. In addition, the Audio Montage's Loudness Meta Normalizer has a number of new presets for adjusting the loudness of multiple clips within a project relative to one another, and also options to normalise peak levels via gain adjustments at either/both the Audio Montage output and Master Section output.

There is plenty to be gleaned across the numerical and visual feedback generated here, and the ease with which peak loudness can be set to your desired target is incredibly useful. No, there is no AI offering you starting points to refine the loudness outcome of your mastering signal chain, but all these new tools do enable you to make more informed mastering choices (or to revisit some mix decisions) if you have a specific loudness standard to hit. Equally, if you are looking to match the loudness characteristics of a commercial reference track, these tools will give you all the feedback you need to steer your processing towards that. This is »

Steinberg WaveLab Pro 12
$499

PROS
- Huge number of new features and refinements.
- Excellent tools for the analysis and management of loudness levels.
- Improved integration with other audio platforms.

CONS
- Deep feature set may feel intimidating for some new users.

SUMMARY

WaveLab Pro 12 is undoubtedly a powerful tool for professional audio analysis, editing and processing. The new features and workflow refinements reinforce its reputation as best in class.

Working via ARA, the full WaveLab Pro 12 editing and analysis experience — including the new Rainbow Waveform display showing here — integrates very smoothly with a suitable host.

» genuinely useful stuff and, for me at least, an absolute highlight of this release.

Hear The Difference

Loudness matching isn't WaveLab Pro 12's only enhancement when it comes to 'hearing the difference', with three new features that might fall into this broad type of functionality. First, within the Audio Montage's Output section of the Inspector, you now have five frequency bands (with customisable crossover points) available that can be soloed during playback. This provides yet another way to compare your own clips to a favourite reference track and I found it particularly useful for evaluating the nature and level of the low end in my own mixes.

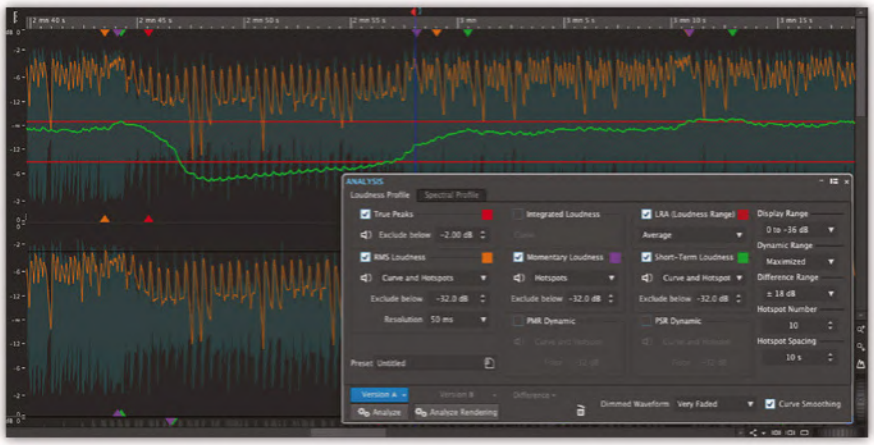
Second, the Audio Montage now has a new Null Test Track feature. Playback from this track is polarity-inverted and then mixed with the Montage output. If you import a rendered version of your track to the Null Test Track, and align it with the unprocessed version, on playback, assuming all the processing that is configured for the unrendered version is functioning correctly, they should cancel, resulting in silence (null). The process is helpful for quality assurance or troubleshooting any problematic plug-ins. The practicalities are made easier thanks to the inclusion of automatic tools to ensure sample-accurate clip alignment.

Third, if you work with multiple studio monitoring configurations, it's now possible to link specific Master Section plug-in chains with specific speaker sets. This might be particularly useful for those using software-based room correction where different EQ correction might be required for each speaker set.

Best Of The Rest

The above selection of personal highlights aside, v12 brings a whole host of other new and improved features, so I'll finish with some additional honourable mentions. For example, the new Rainbow Waveform display mode, while also simply looking cool, provides useful visualisation of the frequency content within a clip's waveform display. This is fully customisable and includes presets (and user presets) for particular tasks such as identifying audio sections dominated by frequencies typical of sibilance, muddy midrange, or with high degrees of low-end dominance.

WaveLab Pro 12 also includes a selection of new plug-ins — Black Wave, VoxComp, Raiser, EQ-P1A, EQ-M5, and SampleAlign — many of which will be familiar to Cubase users. The TestGenerator plug-in has also been revamped. Incidentally, on the plug-in front, as a useful workflow enhancer, the Audio Montage now lets you easily copy settings from one instance of a specific plug-in to other instances of the same plug-in.



The new visual loudness analysis options are also very informative, including the identification of different types of loudness hotspots.

This release also adds options for automatic transient markers, with the ability to snap to these markers when making selections and to navigate through a waveform from one transient marker to the next. Other visual aids include options for automatic vertical zoom on a waveform based upon the waveform's amplitude, synchronisation of scrolling and zooming across multiple audio files and/or Montages, and the ability to zoom in/out while making an audio selection are now also available. New crossfade options improve that area of workflow. For Windows users, a new built-in Steinberg ASIO driver is available. Finally, the CD Window is now termed the Album Window and, amongst the tweaks here, there is now an option to play both track starts and ends, making it easy to audition the gaps between individual tracks as well as their start and end characteristics.

Conclusions

Original developer Philippe Goutier has described WaveLab as the Swiss Army knife of audio processing and editing. It's a perfectly valid analogy but, having reached v12, if you owned a Swiss Army knife that contained as many tools as WaveLab, I suspect you would need several members of a Swiss Army to carry it around. WaveLab Pro 12 is a fully equipped virtual laboratory for the analysis, editing and processing of audio.

WaveLab is used by a number of different user groups — music producers, mastering engineers, sound designers, audio editors for film/TV, forensic audio experts and more — and these established users will be able to draw their own conclusions about where the value lies to them in the impressive list of refinements. However, I do think a few additional words aimed at potential new users are worth making. WaveLab Pro 12 is undoubtedly a powerful, feature-rich, audio laboratory. However, while Steinberg have continued to do a great job in making the workflow efficient, things can — and do — get deep pretty quickly. That's not a criticism of any sort; it's simply a statement of the reality. If you want to have surgical control over almost every aspect of your audio, you need a compressive and surgical toolkit. WaveLab Pro 12 provides exactly that; it's powerful and feature-rich, but it pulls no punches.

Personally, I think that's a very good thing. Yes, designing software that makes specific complex tasks accessible to any user level certainly has its place, but it requires considerable skill on behalf of the developer if it is also to satisfy more experienced users. While the workflow enhancements in this new WaveLab release are all welcome, hats off to the development team for resisting the temptation to dilute the WaveLab experience. For some tasks, surgical, powerful tools are still needed and, in this reviewer's opinion, WaveLab Pro 12 is simply the best professional audio editing environment on the market by a considerable distance. ■■■

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Cinesamples Musio

Virtual Instrument Catalogue

Cinesamples' subscription package promises an awful lot of bang for your monthly bucks.

JOHN WALDEN

You can browse the full Musio catalogue within the plug-in while the Instrument Rack shows all currently loaded sounds on the left side of the UI.

Cinesamples' extensive catalogue of Kontakt-based orchestral sample libraries will be very familiar to those who inhabit the world of film and TV composition. They create premium libraries suitable for high-profile projects and, as with the obvious competition from Native Instruments, EastWest, Spitfire and others, they are also priced accordingly. For example, one of Cinesamples' most popular orchestral strings libraries — CineStrings Core — will currently cost you upwards of $400.

Understandably, therefore, building a complete orchestral palette via individual Cinesamples Kontakt libraries represents a significant investment. Or maybe not... because Cinesamples have now introduced Musio. This platform provides access to a substantial (and growing) selection of their sample libraries for either a $9.99 per month subscription or a one-off payment of $399. So, what exactly is Musio?

Cinesamples Musio
From $9.99 Per Month

PROS
- Huge array of high-quality sample-based instruments.
- Efficient, streamlined workflow.
- Impressive value.

CONS
- Might be too streamlined for some users?

SUMMARY
The value offered by Musio's catalogue of high-quality sounds is undeniable. If you are happy to embrace the streamlined design ethic of the plug-in, it's an attractive proposition.

Top-tier Sounds

The Musio platform combines two key elements; a catalogue of sample libraries and a plug-in front end to access them with. In order to keep this review at a reasonable length, let's address the first part of the Musio equation — the catalogue of sample libraries — by saying something simple and on-point; the libraries themselves sound fabulous.

Within the current Musio catalogue, you get access to a huge selection of the CineBrass, CineString, CinePerc and CineWinds titles, the various Drums Of War libraries, the Men and Women Of The North vocal libraries, a whole range of the more specialist titles and, currently exclusive to Musio, a series of 'Artist' titles and some more contemporary libraries covering classic synth and drum-machine sounds. Cinesamples maintain a complete list of what's currently included on their website and for orchestral/hybrid orchestral composers, you could easily create a complete score without stepping outside the sounds Musio has to offer. And, since Musio's launch, Cinesamples have been gradually porting more of their individual libraries over to the Musio format, so this is a catalogue that is undoubtedly going to expand over time.

Access All Areas

Musio is actually available in two forms; Musio and Musio 1. The former is the subscription model and currently priced at $9.99 per month (or $99 per year if purchased annually). An active subscription gives you unlimited access to the existing libraries via the Musio plug-in front end, as well as access to any new library additions, and all updates to the plug-in itself.

In contrast, Musio 1 is available for those that prefer a one-off payment. For $399 (less than the price of many of the single Kontakt-based libraries) you get perpetual access to all the Musio libraries available at the time of purchase as well as lifetime updates to the Musio plug-in. With either Musio or Musio 1, you get an initial 30-day trial period for free.

Whichever way you go, Musio undoubtedly offers an amazing amount of sample content at a modest price.

The Front End

However, one key point is worth noting. Access to the sounds is exclusively through

the Musio plug-in rather than Kontakt, and there are some notable differences. The Musio plug-in (AAX, AU and VST3) offers four key functions; instrument browsing/auditioning, sample downloading, sample management (in conjunction with a further app — Musio Connect — but they work seamlessly together) and sample/instrument playback. As shown in the screenshots, this is all done via a modern UI with a very streamlined design.

The Explore (focused on new library additions) and Catalogue (a comprehensive list of all the currently available titles) tabs provide alternate ways to browse the library catalogue, while the very useful Search option (located top right) lets you type in your search term and Musio will provide a list of all the individual instrument patches that match from within the entire catalogue.

From any list of instruments, you can audition a brief example (useful to find exactly the sound you are looking for) and then download it to local storage. The Load option will also download the instrument's samples if needed and, once done, the instrument will appear within the Instrument Rack on the right of the UI. Yes, a speedy Internet connection is useful but, that aside, this is a very simple and streamlined process. The Musio Connect app provides additional options for handling things like local storage details and provides a list of all the instruments currently downloaded. Incidentally, one Musio feature requested by early adopters has been the additional option to download a complete library with one click; at the time of writing, that's apparently something Cinesamples have in development.

Like Kontakt and Opus, Musio can be multitimbral and, for each instrument, you can specify both the MIDI channel and the output channel (provided your DAW supports multiple outputs). The Instrument Rack also supports presets so you can easily save/recall your own instrument configurations.

However, perhaps the most striking element of the Musio's UI is the simplified control set available for each individual instrument once loaded. This minimalist approach is most certainly a deliberate design choice by Cinesamples; you are given just enough options to shape the tonality, dynamics and ambience applied to the sound, but without the myriad options you might find in a typical Kontakt-based instrument (including some of these same Cinesamples instruments in their Kontakt format). The Musio concept is designed for those composers who just want to load their sounds, fine-tune the most important properties (or not), and then get writing. No fuss, no distractions, just high-quality sounds you can get to work with.

The streamlined design ethic favours a compact control set for individual instruments but there is enough control to tailor sounds to your needs. Key parameters are linked to the mod wheel by default but others can easily be assigned to MIDI CC numbers.

Those tweakable options that are included vary by library. Importantly, for many of the orchestral libraries, this does include keyswitchable articulation presets (including a neat system for customising the key layouts), but there are also some things that some users might think are a little too streamlined. For example, whereas the Kontakt-based Soundscapes library provides you with options to change the way the sample playback engine manipulates the underlying sample used in a preset, in Musio, you don't interact with the sample at such a deep level, although there are useful sound-shaping options. The same is true of the exclusive-to-Musio synth sounds. These don't provide a full virtual synth engine with oscillator controls; samples provide the sound of the original oscillators and Musio then lets you adjust some filter, envelope and effects options (similar in concept to IK Multimedia's Syntronik). It works well but it might not appeal to those that like to program their synth sounds from scratch.

That said, it's clear that Cinesamples have a development program laid out for Musio and certain features have already been added since the initial launch. Cinesamples have also announced that advanced mixing support is also on the way, supporting multiple microphone positions for libraries with a suitable sample base. The feature list will undoubtedly continue to expand but, hopefully, without compromising what is, in use, a very attractive and streamlined workflow.

One further feature is also worth highlighting. If you regularly work with collaborators also using Musio, sharing projects becomes incredibly easy. Whatever selection of instruments you might use in creating the project will automatically be available to your collaborator even if they have not yet downloaded the specific presets. For lead composers employing a writing assistant or two, this could be a massive workflow benefit.

A Bigger Bang?

There is undoubtedly a debate to be had about the merits of subscriptions versus one-off perpetual purchases. However, if your music is part of what puts food on the table, as an investment in your business, Musio's $9.99 per month is probably less than you would be paying for your phone contract, your fibre-based Internet access, or even a moderate coffee habit. As a composer, I'm not sure I can think of a bigger bang for your buck than a product like Musio might bring.

Of course, as a one-stop virtual instrument shop, Musio is not the only game in town. If you prefer a one-off purchase for a 'mega-bundle' of virtual instruments (and plug-ins), Native Instrument's various Komplete bundles (see the December 2022 issue for a review) provide an alternative to Musio 1. For the subscription route, the most obvious comparison is EastWest's Composer Cloud. Both of these alternatives are very impressive, but Musio brings a distinctive streamlined workflow that will, I suspect, appeal to a broad audience. Does that include you? Well, that free 30-day trial is there precisely so you can find out. Musio is a pleasure to use and on a bang-for-buck basis it's an undeniably impressive package.

$ Subscriptions from $9.99 per month, Musio 1 perpetual access $399.
W www.cinesamples.com

Pulsar Audio Primavera

Spring Reverb Plug-in

With an array of different modelled devices and controls you won't find on the original hardware, have Pulsar breathed new life into the humble spring reverb?

NEIL ROGERS

Since their first release in 2017, French software developers Pulsar Audio have been steadily building an impressive collection of plug-ins, with emulations of esoteric hardware effects and processors that, in the hardware world, would be out of reach for most of us. Regular readers will know that I have a real fondness for hardware spring reverbs, but in some scenarios plug-ins are more convenient, so I was very happy to check out Pulsar's latest release.

Overview

Primavera promises us a smorgasbord of spring reverb options based on some classic and often lesser-seen devices: Pulsar say that it features precise emulations of both the electronic and mechanical behaviour of six classic spring reverb units. These include the Pioneer SR202, Hawk HR12 and Roland RE-201 as well as more broadly descriptive options they've called Twang and Tubby. A final option is GBS, which I assume is based on the Great British Spring, a legendary design that was housed in a sizeable piece of drain pipe!

We're also treated to a plethora of modern control options with which to shape the behaviour and tone of these virtual 'springs', and this promises to make the plug-in more flexible than any of the modelled hardware units would be when mixing. A quick tour of the 'unit' from left to right begins with a preamp section. This offers us the option of driving the input into a virtual valve or germanium preamp, and there's the welcome addition of simple high- and low-pass filters that affect the wet signal only. Over on the right, a wet/dry blend control allows you to set the plug-in to only use this preamp stage, only the reverb, or anything in between. The centre section presents a choice of the six emulations, along with the more typical software reverb controls for pre-delay, decay and width. Less conventional are the controls for Tension, which allows us to stretch or loosen the physical model of the springs, and Excitation, which allows you to dial in more or less of the more explosive effects of physically hitting a spring hardware unit! A handy ducking facility allows you to rein in the effect while there's an input signal. There are also useful oversampling options, with the ability to specify different settings for real-time use and for an offline render, and a drag-to-resize GUI, which is nice.

Lo-fi To Hi-fi

I managed to get Primavera installed quickly, during a short break in a live recording session I was engineering, and getting a workable sound up and running for a clean electric guitar track before the band came through to listen to a take was a breeze. The Twang setting is based on a reverb found in Fender guitar amps and

Pulsar Audio Primavera
$105

PROS
- Six flavours of virtual spring to choose from.
- Fun and creative to use.
- Extra control features allow for finer mix placement.
- Preamp section allows for added 'grit' and character.

CONS
- None.

SUMMARY
This plug-in features a selection of convincing spring reverb emulations and a fun, well thought-out selection of controls that expand the versatility of this much-loved style of reverb.

it certainly sounded convincing when used on clean or slightly overdriven guitar recordings.

Clearly, Pulsar have taken a lot of care to make this plug-in easy to use, and I was particularly impressed at how 'natural' it felt when I skipped between the different flavours of spring. They've organised the six options in such a way that they range from 'lo-fi' to 'hi-fi', with the general idea that you start with the more 'obviously spring' effects that most users will be looking for, and things then get a bit cleaner and more versatile (as you might want for vocals, for example) as you step through the options. I often found myself favouring the RE-201 and GBS as a starting point that suited my personal taste. The HR12 and Tubby sound darker and more dense, and these seem to work great for drums, some synth sounds and, on occasion, even a bass guitar.

As great as hardware spring reverbs can sound, and as fun and inspiring as they can be to play with, they can sometimes be a bit frustrating to use. That's due to the difficulty in physically engineering control parameters to change decay times and the like. Primavera dispenses with the constraints of the physical world, offering a great selection of tools for fine-tuning the result, without it ever really feeling like you have too much choice, and the big one, sitting smack bang in the middle of the GUI, is the Decay control. This enables you to transform any of the reverb options into a more expansive effect, or to tighten things right up to the point where the springs are more of a tone-shaping tool than a spatial one.

The width and pre-delay sections perform exactly as you'd expect on a software reverb, and the Tension and Presence controls offer the opportunity to really fine-tune the tone and character of your chosen spring model. I reckon we could all do with an Excitation control in our lives! This intriguing control allows you to dial in more or less of what I would describe as the 'boing' of a spring reverb. You can also get into more special effect territory by clicking on the CGI springs, which are animated as they respond to this virtual 'kick'. The ducking effect can also offer a way to make this effect useful on the right sound. Of course, there does come a point where you can dial things back too much — at this stage, you should probably ask yourself if you might like to try a different style of reverb altogether — but the additional controls certainly felt very useful to me in those closing, really detailed stages of a mix.

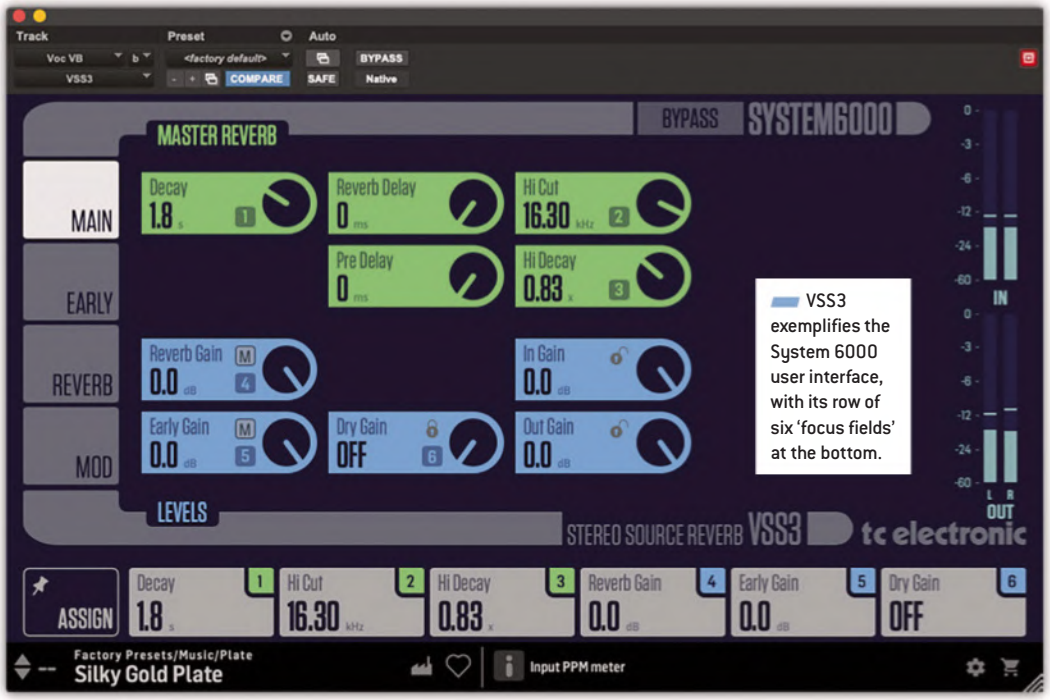
Wrapping up my whistle-stop tour of Primavera with a brief mention of the preamp section: you have the option of dialling in more 'grit' using the tube or germanium transistor-based preamps I mentioned above. You can go pretty crazy with this, or just add a touch more character if needed, and can use the wet/dry control to give you just the preamp (no reverb) if you want to use it as a saturation/distortion effect. While most of us will have preamp emulations of some sort or another in our plug-in folders already, it was interesting to hear how this stage seemed to interact with the different reverb styles: it felt to me like there was more going on here than when you just drive any preamp before a reverb.

Summing Up

Primavera was my first encounter with Pulsar's software, and I was very impressed: it has been a wonderful addition to my software effect options, and it ticks a number of my 'what I want from a plug-in' boxes. You can slap it on a track during a writing or recording session to get a bit of instant 'vibe', whilst also having the flexibility to fine-tune and sculpt your sounds towards the latter stages of a mix. I either own or have owned most of the units that are emulated in this plug-in and whilst I do think that the hardware still offers a little 'something' that this doesn't quite manage, these are excellent-sounding emulations — better than most I've heard and convincing to the ear — and there are more than enough modelled units to choose from, making it both enjoyable and convenient. The additional control options then take this plug-in to another level: as well as a fun 'colour box', it's also a very serious mixing tool, as it's much less of a fight to tuck your chosen colour into a mix. If you're into your lo-fi and characterful-sounding effects you should definitely take the full-featured 14-day trial of Primavera for a test drive. ■■■

$ €99 (about $105). Discounted to €59 ($63) when going to press.
W https://pulsar.audio

TC Electronic System 6000 Native Bundle Plug-in Collection



VSS3 exemplifies the System 6000 user interface, with its row of six 'focus fields' at the bottom.

Is there still a place in our plug-in folders for the core algorithms from TC Electronic's legendary System 6000?

SAM INGLIS

When I joined the staff of *Sound On Sound*, sometime in the 18th Century, there were three big names in digital effects: Lexicon, Eventide and TC Electronic. And, at the time, they were all busy spinning the technology from their high-end studio processors out into more affordable units. In the case of Danish DSP wizards TC Electronic, that gave us project studio staples like the Triple C dynamics processor and M One effects unit, and perhaps most ubiquitously, the Finalizer range of mastering processors. And TC had their fingers in many other pies too, for example with rackmounting guitar effects like the Gforce, not to mention the Dice II chipset that powered many FireWire audio interfaces.

As the market for digital studio hardware dwindled in the 2010s, new opportunities arose elsewhere. Now part of Music Tribe, TC currently make a huge range of guitar and bass effects pedals, floorboards and bass amps. But the studio expertise that powered iconic products like the System 6000 hasn't been neglected. Indeed, several members of the team that developed those products are still active at TC, including studio product specialist Christian Gadegaard Frandsen.

Back in the day, TC were quick to spot the potential of the Pro Tools TDM system, and eventually introduced their own Powercore DSP co-processor. Both platforms allowed engineers mixing 'in the box' to employ TC signal processing algorithms as plug-ins, rather than routing the signal out of the computer to a black box in a rack. Fast-forward to 2024, and there is now a comprehensive range of TC plug-ins in all the major native formats, available individually or in themed bundles. Some are entirely new, some are essentially ports of those classic algorithms, and some are faithful emulations of even older TC hardware products. The 'one ring to rule them all' is the TC Studio Elite Collection, which includes a total of 20 plug-ins encompassing seven themed collections. Among them is the System 6000 Native Bundle, the subject of this review.

Feeding Of The 6000

The TC System 6000 was TC's flagship digital processor. A rackmounting unit with a separate remote controller called the Icon, it was offered in several 'mainframe' configurations to suit particular use cases, with additional algorithms available to buy separately. For example, you could buy it as a Reverb 6000, pre-loaded with reverbs and other effects, and then later upgrade to full System 6000 specs by adding dynamics, EQ, noise reduction and other algorithms; or you could opt for the Mastering 6000, which shipped with various dynamics and EQ capabilities but could likewise be upgraded. Some individual System 6000 algorithms were also available as plug-ins for TDM and Powercore systems.

Although there were all sorts of effects and processes available, the Virtual Space Simulation reverbs and multiband dynamics were the jewels in the System 6000's crown, and the five plug-ins that form the System 6000 Native Bundle repackage those key algorithms. There are three reverb plug-ins — VSS3 Native, VSS4HD Native and Nonlin2 Native — plus the MD3 Native and MD4HD Native mastering processors. All major native plug-in formats on macOS and Windows are supported, and authorisation is to an iLok account.

The native plug-ins are, as far as possible, faithful to the implementations that appeared in the original hardware and Powercore/TDM plug-ins, which in turn means they largely replicate the user interfaces as they appeared on the Icon controller's touchscreen. This used

a control paradigm whereby parameters were selected for editing by touching an on-screen button, then adjusted by moving a fader. Consequently, even on/off switches and settings chosen from lists appear as 'rotary' controls. Retaining the System 6000 interfaces also means these plug-ins don't have the visual bells and whistles you find in many modern rivals. For example, many multiband processors display the crossover frequencies and other parameters graphically over an FFT display of the incoming signal, but that's not an option in MD3 or MD4HD.

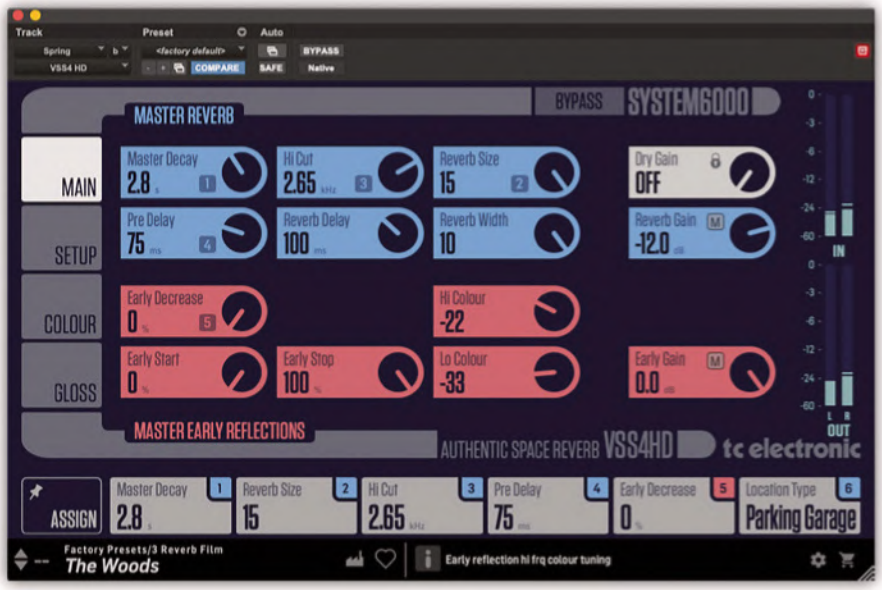
Considering its age, though, I actually think the user interface paradigm has held up rather well. These are relatively complex plug-ins, with at least 20 parameters exposed and often many more, yet they never feel overwhelming or unclear. The controls are sensibly distributed across multiple tabs, and the freely resizeable windows can be made compact enough to fit comfortably on any laptop screen, or large enough for the most ginormous monitor. These aren't the sort of plug-ins where you can randomly drag pretty graphics around and hope for the best, but as long as you have some understanding and experience of the basic processing types and parameters, they'll be straightforward, logical and fast to use. There's also a commendable consistency of design between all five plug-ins in this suite, meaning that once you've learned one of them, the others are easy to pick up.

A good example of this is the six 'focus fields' that appear along the bottom of each plug-in window. These remain visible whatever editing page you're on, and can be assigned to any control you like. So if, for example, you typically do most of your reverb editing using four or five parameters that happen to be spread across different editing panes in VSS3 or VSS4HD, you can drag these to the focus fields and shape your sound without the need for constant page flipping. You can also choose to 'pin' these assignments so that recalling a preset doesn't change them.

V Is For 'Verb

In the days when TC and Lexicon were the only serious players in digital reverb, they were regarded as having slightly different strengths. If you wanted something larger than life to add a touch of glamour, you would perhaps favour Lexicon, but if you wanted a pristine, clear sound that could pass for natural rather than artificial reverb, you'd plump for TC. In practice, this perception underplays the capabilities of both; and what's surprising about VSS3 and VSS4HD is how well they stand up against modern reverb plug-ins. The latter, in particular, is definitely not some period piece you'd wheel out on the off-chance someone wanted a retro '90s mix. It's a seriously powerful and versatile reverb plug-in that can produce everything from tight ambiences to expansive halls.

VSS features that were perhaps novel and exciting at the time, such as true-stereo operation, are now commonplace, but the implementation here still works very well. Like many algorithmic reverbs, VSS4HD is based around separate engines generating the early reflections and the reverb tail.



Billed as an "authentic space reverb", VSS4HD set new standards in replicating realistic acoustic spaces.

TC Electronic System 6000 Native Bundle

$1095

PROS

- Still immensely powerful and can sound great.
- Authentic System 6000 algorithms and presets.
- Clean, efficient user interfaces.

CONS

- No loudness metering in MD3 or MD4HD.
- Limited graphical feedback compared with some plug-in alternatives.

SUMMARY

TC's System 6000 algorithms were state of the art when they were introduced, and they've turned out to be ageless. Audio professionals who are willing to invest the time and money will never outgrow the possibilities of these plug-ins.

The Main page provides extensive and independent control over the parameters for both. Setup is where you choose a room type from a list of 12, and position the left and right sources within it, while the Colour pane offers detailed control over the tone of the reverb. Finally, Gloss is where you apply modulation to the tail. If you've ever used a conventional studio reverb, it'll all feel very familiar, and for good reason: this is one of the products that defined those conventions in the first place!

In some ways, though, I actually found VSS3 the more useful of the two. Although VSS4HD is still remarkably effective at creating smooth, rich virtual halls and chambers, it didn't displace existing favourites of mine such as Liquidsonics' superb Cinematic Rooms. VSS3, by contrast, has a distinctive character that I haven't heard in another plug-in. It's not lo-fi in the sense of being grainy or gritty, but it has a brasher and more mid-focused sound that's often perfect on sources such as vocals. For some reason, its output is typically several dB higher than that of VSS4HD, but even if you match their levels, VSS3 sounds louder and more 'in your face', if that's something you can say of a reverb plug-in. It's versatile enough that you could use it on almost anything, yet distinctive enough that you couldn't easily substitute it with a generic reverb.

Nonlin2, by contrast, was always intended as a special effect reverb, and still excels in that role. The idea behind a non-linear reverb is that the usual decay pattern is disrupted or even reversed, so that the reverb builds up rather than dying »

Nonlin2 specialises in the production of unnatural and weird reverbs.

» away. The classic application is on drums, but it's by no means limited to cliched '80s gated snares. A variety of short algorithm types is available, from emulations of vintage AMS units and the like to the descriptively named Box and Hollow, and some effective ambiences. There's no separate early reflections section; instead, there's a graphical Envelope not unlike a synth ADSR, with an Attack time that sets how long the reverb takes to develop, and a Hold parameter that forces it to remain at a constant level for up to 500ms before the release phase kicks in. The factor that sets this plug-in apart, though, is the modulation. You can choose from a wide variety of Twist options with names like Aircon, Kazoo and Radiator, which warp the sound in a variety of entirely unrealistic but fun and rewarding ways. Many reverbs have come and gone since the System 6000 was developed, but I honestly can't think of another plug-in that works or sounds like this one.

The Final Countdown

As the names suggest, the relationship between MD3 and MD4HD mirrors that between VSS3 and VSS4HD; they represent successive generations or iterations of the same idea. Both are, in essence, complete mastering processors, and it was the MD3 that formed the heart of the hugely popular Finalizer back when we were all discovering the joys of home mastering. Once again, it's tempting to wonder why you might need MD3 when MD4HD is a 'better' version of the same thing, and once again, it turns out that the older and supposedly lesser version still has something to give.

For one thing, I would wager that at least half of the records made through a TC Finalizer simply used one of the presets. Unlike the System 6000, these 1U rack units were not blessed with a touchscreen remote control, and although a Wizard function made editing easier, you were still dealing with a parameter wheel, a small screen and lots of abbreviations. The MD3 plug-in includes many of the most popular presets from the hardware Finalizer models, and thus offers 'that sound' but a much more user-friendly interface. It implements the instant mastering settings for which the Finalizer was renowned, but makes it a much more practical proposition to go in and tweak them.

However, it's an oversimplification to say that the MD3 plug-in is simply a software Finalizer, because both units had several processing sections, which are not all identical. The biggest contributor to the MD3 'sound', if there is such a thing, is the three-band compressor, and this is exactly the same algorithm as found in the Finalizer. Any multiband is only as good as the crossover filtering, which can introduce unpleasant phasey artefacts if done wrong. Despite its age, the MD3 algorithm is still extremely transparent. The crossover points are not continuously variable, but there are sufficient options to cater for most needs, and control over the compression parameters is extensive, with separate threshold, ratio, attack and release times and make-up gain for each band. What's even better is the provision of various linking options that allow multiple related parameters to be quickly adjusted together.

The compressor is preceded by a separate three-band expander, which is perhaps less useful in this day and age than it was when the Finalizer was launched; there's a lot less tape hiss to clean up nowadays, and a lot more dedicated noise-reduction tools around. MD3's four-band EQ is likewise not state of the art any more, but is sonically blameless, and it's ideally configured for its role towards the start of the chain, allowing you to drive different regions of the frequency spectrum more or less aggressively into the dynamics. (It's not possible to change the fixed processing order.)

The output stage is where MD3 and the Finalizer differ most. The Finalizer had »

The MD3 plug-in reproduces much of the functionality of TC's Finalizer hardware units.

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» a multiband limiter to complement the expander and compressor, whereas MD3 has a full-band, brickwall limiting stage. Nevertheless, this very much holds its own against newer mastering limiter plug-ins. In particular, it's capable of true peak limiting, meaning you can guarantee that your mix will be free of intersample overs that can cause clipping distortion on some playback systems. There's also adjustable release time and a Soft Clip option — two, in fact, since there's a separate soft clipping stage that can be introduced at the start of proceedings. Both MD3 and MD4HD can be operated in Mid-Sides mode, which opens up a whole new can of mastering possibilities/worms, as does the Detector Crest dial, which allows the side-chain detector to be set to peak, RMS or various inbetween options.

One MD3-only feature to be careful of is the button labelled Auto Makeup Gain. This is enabled by default, and operates pretty aggressively; if you're wondering why reducing the thresholds or increasing the ratios is making everything louder rather than quieter, this is likely to be the culprit. Used with care and good taste, MD3 is a hugely capable mastering processor that, compared with modern rivals, lacks little apart from graphical sophistication. And used with indulgence, it will transport you back in time two decades!

The Four Factor

Despite the name, MD4HD offers five bands of dynamics processing to MD3's three, and there are more EQ bands and filter types, too. There are also plenty of other differences. A key MD4HD feature is parallel processing, which is available in three flavours. By default, Parallel mode blends the dry and compressed signals in equal parts, with a -6dB gain offset; the Wet Gain control can be used to vary the balance. Parallel UL mode is the same, except that there are two unlinked side-chain channels, which is particularly useful in Mid-Sides operation but can add more 'body width' even in L-R stereo. Finally, DXP mode is 'true' upward compression, boosting signals below the threshold but leaving peaks alone. Even in standard multiband compression mode, each band adds Defeat Threshold and Defeat Ratio controls to the usual compressor parameters, allowing the bands to revert to unity gain with quiet signals that fall below the Defeat Threshold. This helps to avoid low-level noise being raised too much. Another area where

The specialities of MD4HD include parallel and upward compression, enabling it to control dynamics in a very transparent way.

MD4HD expands (if you'll pardon the pun) on MD3 concerns the output limiter, which features several different Profiles, with names like Soft, Dynamic and Loud, and the option to unlink the two channels.

MD3 was already highly versatile, but the extra bands and the various parallel and upward compression options on MD4HD open up many new possibilities. These are aptly exploited by the presets (including a number created by mastering engineer Bob Katz), which is lucky, because setting up the entire plug-in from scratch would be a fairly daunting prospect. Whereas it would be feasible to start with a 'blank slate' in MD3, there are simply so many options and parameters in MD4HD that you're better off picking a suitable starting point and adapting it to taste. At first, even this can sometimes be confusing, in that changing parameters doesn't always have the effect you might expect, but you quickly learn to check key settings such as the compression mode on loading a preset.

> "You can push and pull a mix in any direction you like, and it never seems to lose its sonic integrity."

Sonically speaking, MD4HD doesn't really have 'a sound'. Quite the opposite, in fact: its speciality lies in processing audio in such a way as to completely obscure the fact that any processing has taken place. You can push and pull a mix in any direction you like, and it never seems to lose its sonic integrity, even when you're slamming the output limiter or adding scary amounts of top end. Its transparency is remarkable, and although there's a fairly considerable learning curve involved in getting the best from it, it's a curve well worth following. The MD4HD algorithms may be 'vintage', but if there are newer products that do things better, I haven't heard them.

Quality is always in fashion, and both MD3 and MD4HD are every bit as useful in sonic terms as they ever were (with the possible exception of the expander stages). Indeed, once you get used to the user interface, the fact that it forces you to use your ears instead of watching an FFT display dance around is often a good thing. But there are a couple of ways in which these plug-ins could benefit from being modernised. Neither offers output dither, so you'll need to follow them with a separate plug-in when you render a 16-bit output; and, more importantly, there's no loudness metering, so it's not easy to work towards a loudness target in LUFS.

The System 6000 was always a flagship product, and that's reflected in the pricing of the Native Bundle. These are serious plug-ins aimed at a professional market. They're supremely versatile, and in the hands of someone who knows how to use them, can deliver impressive results with a speed and efficiency that's still hard to match. TC's algorithms set new standards at the time, and more than 20 years later, they remain right at the top of the tree. ■■■

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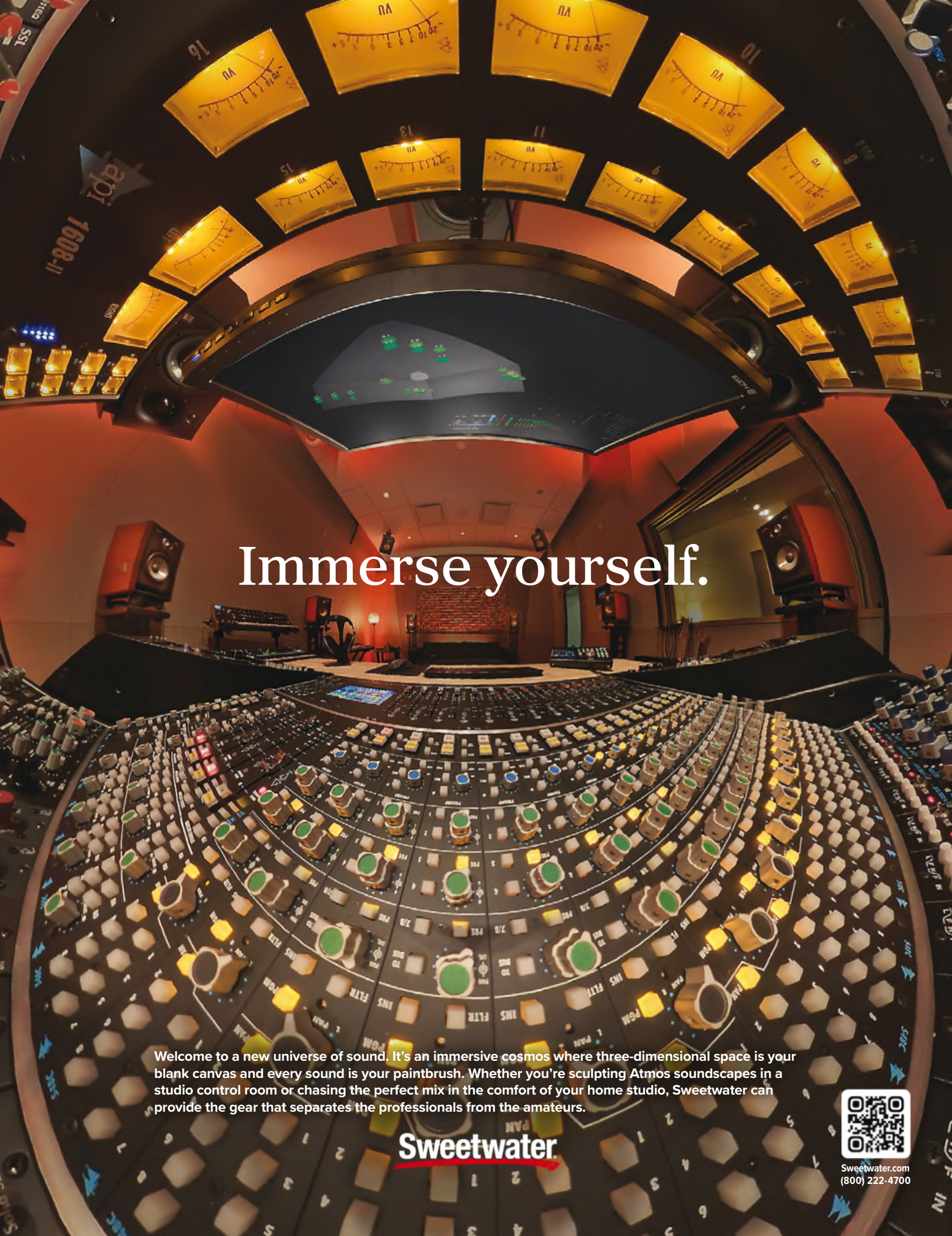
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Barefoot Footprint03

Active Monitors

These compact three-way speakers are Barefoot's most affordable monitors to date.

PHIL WARD

Somewhat surprisingly, I've not reviewed a Barefoot monitor since the Footprint01 in 2017. Back then, Barefoot were still relatively new entrants to the monitor market, but these days they feel like part of the establishment, and the company should be applauded for surviving and apparently thriving in an intensely competitive market.

With the Footprint03, Barefoot have in two respects entered new territory. Firstly, the Footprint03 is significantly more compact and affordable than other products in the range. And secondly, it's Barefoot's first reflex-loaded monitor. All previous Barefoot monitors have been closed-box designs, and the company have been evangelical advocates of the benefits of closed-box loading. It will be fascinating to see how the engineers at Barefoot have managed the trade-offs inherent to reflex loading.

Baby Steps

The Footprint03 is a three-way active monitor of relatively compact dimensions and light weight that ought to be easily compatible with the majority of studio installations. It's an attractive and neat-looking design that follows the established Barefoot aesthetic of dark grey textured paint, rounded cabinet edges and aluminium trim panels.

The rear panel offers just mains power and balanced analogue input sockets, along with a stepped input-sensitivity knob. There are also a couple of tapped, standard pitch-mounting hardware attachment points, and I can absolutely see Barefoot having multi-channel Atmos studio aspirations for the Footprint03. Why sell monitors by the pair when there's applications out there for seven, nine, 11, or even more? The Footprint03 is optionally compatible with Barefoot's proprietary MEME (Multi Emphasis Monitor Emulation) technology, which aims to offer the subjective characteristics

Barefoot Footprint03
$1995

PROS

- Compact and well conceived overall design.
- Revealing and explicit midrange.
- Well judged reflex-loaded bass.
- Generally well balanced, high-performance monitoring.

CONS

- Slight tonal quirks.

SUMMARY

With the Footprint03, Barefoot Sound have in some respects reinvented themselves as manufacturers of compact, affordable monitor speakers, and they've done so very successfully. This is a highly capable and competitive monitor that deserves to succeed.

The rear panel hosts a single combi XLR/jack input with a stepped input gain

of alternative monitors, and the rear panel offers the necessary sockets. However, the review pair didn't include the MEME hardware controller so I wasn't able to try it.

Downstream of the input, the Footprint03 includes separate Class-D amplification for its bass, midrange and tweeter drivers, rated at 100W for the bass and midrange and 60W for the tweeter. Prior to its amplification stages, the Footprint03 incorporates a DSP module, with 28-bit/96kHz conversion, which handles EQ and crossover functions. However, there's an interesting twist in the way the Footprint03 crossover filters are configured, in that an initial low-frequency filter is implemented in the analogue domain before the DSP. Once the low-, mid- and high-frequency signals reach the DSP, further digital filter stages then complete the necessary crossover filter function. This patent-pending composite filtering technique, which Barefoot call SPOC (Spectrally Optimised Conversion), is claimed to ensure that digital resolution isn't lost in accommodating the typical spectral balance of music, where high frequencies generally reach lower maximum levels than low and mid frequencies.

Motoring Along

The drivers comprise a 165mm aluminium-cone bass unit, a 90mm aluminium-cone midrange unit and a newly revised, 25mm fabric-diaphragm ring-radiator tweeter that, say Barefoot, boasts increased sensitivity and diaphragm excursion capability and reduced compression. The ring-radiator tweeter architecture is fundamentally that of a dome tweeter, except that the apex of the dome is fixed and the surround area outside the voice-coil diameter is enlarged. Ring radiators offer a slightly different set of characteristics and compromises compared to traditional dome tweeters, and when they were first introduced (by Danish driver specialist Vifa) in the late 1990s, they made quite an impact among driver engineers. The impact has died down in the intervening couple of decades, but Barefoot have »

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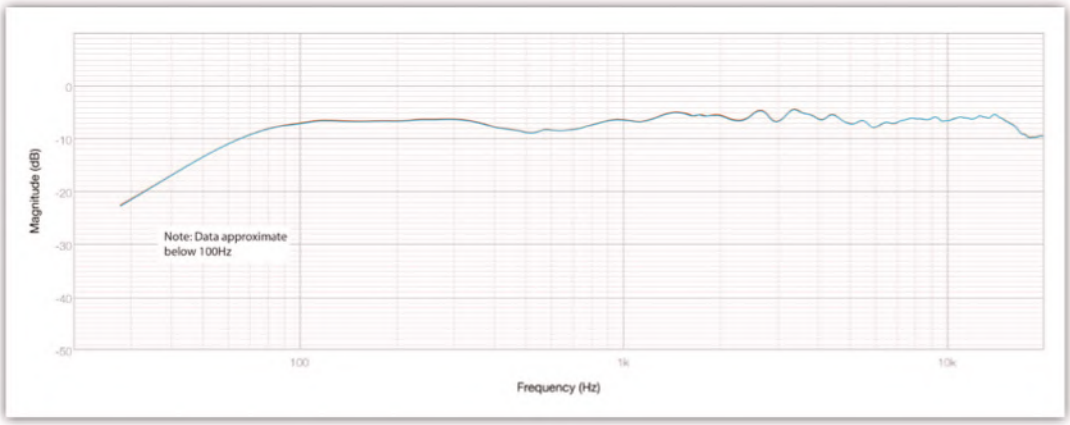

Diagram 1: On-axis measurements of the Footprint03's frequency response, measured at both 84 and 94 dB SPL (at 1m). The results are normalised and overlaid — and yes, there are two traces in this graph!

» stuck with the ring-radiator format, and to my mind, it definitely offers some benefits. A significant one is typically a very low fundamental resonance frequency, combined with good high-frequency directivity.

The midrange driver is of small diameter, so you would expect it to be relatively limited in terms of low-frequency bandwidth. This will tend to pull the bass-midrange crossover frequency upward, and that's confirmed in the Footprint03's 600Hz specification. The bass-midrange crossover in a monitor with a larger midrange driver would typically be around an octave lower. At the other end of the midrange band, the driver's small size takes the pressure off the tweeter by enabling a higher-than-usual 3.9kHz high-frequency crossover frequency. This will potentially minimise distortion at the lower end of the tweeter band, and reduce the possibility of thermal compression — a phenomenon which typically affects tweeters first because the thermal mass of their voice-coils is smallest.

A small midrange driver only requires a modest isolating enclosure, and this helps keep the overall monitor dimensions compact. The driver features a small roll-surround, its minimal size made possible by the diaphragm not requiring the generous excursion levels of bass drivers. Having said that, Barefoot specify the midrange driver as having ±3.5mm linear excursion potential, which is pretty generous for such a small driver. The driver incorporates multiple copper shorting rings in its motor system, which bodes well for distortion performance and subjective clarity.

The Footprint03 bass driver features a generously dimensioned rubber roll-surround and is specified with an impressive ±9.5mm linear excursion. Like the midrange driver, the bass driver's motor system incorporates multiple copper shorting rings for lower distortion and minimal inductance modulation (shorting rings significantly reduce the degree by which a driver's voice-coil inductance varies with diaphragm position). The driver appears to be a very competently designed example, although you can't entirely separate a bass driver from its enclosure, and with that of Footprint03 being reflex-loaded via a side-mounted port, there's a few things to unpick.

It's All In The Reflexes

The decision in monitor design between closed-box or reflex (or even transmission line) loading isn't entirely one of electro-acoustics. Sales and marketing factors play a part too, because the size and manufacturing cost of a monitor are fundamentally influenced by its bass loading technique. And with the Footprint03 being compact and relatively inexpensive, reflex loading is no great surprise. I'll explain why that's the case.

The choice of reflex or closed box falls out of the more fundamental trade-off inherent to any moving-coil speaker between cabinet volume, electro-acoustic efficiency (acoustic power out versus electrical power in) and low-frequency bandwidth. Change one of those factors and the pesky laws of physics inconveniently demand that one or both of the others must change also. For example, if you want more low-frequency bandwidth, either the efficiency must fall or the cabinet volume must grow (or both). However, if you slip reflex loading into the equation, you can persuade physics to let you squeeze a little more out of the fundamental sum. That's possible because what's actually happening with reflex loading is that the acoustic energy radiating from the back of the bass driver diaphragm is being used to drive a reflex port resonance, rather than being dissipated as heat inside the cabinet. So with reflex loading you can potentially reduce the size of the cabinet without

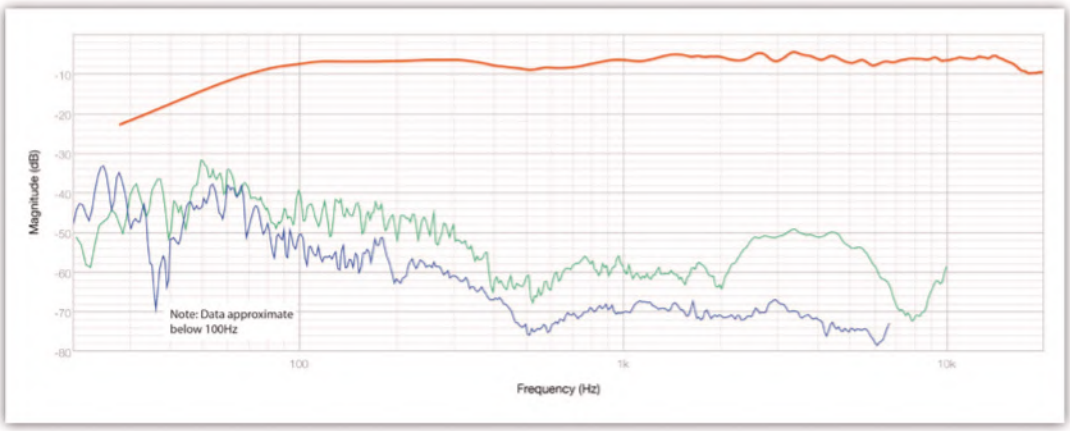

Diagram 2: The axial frequency response (94dB SPL at 1m), and second- and third-harmonic distortion (green and blue traces, respectively).

losing bandwidth or efficiency. And even though you might imagine you've cheated the laws of physics, you still pay a price; it just lives in the time domain, where there's an unavoidable latency inherent to the 180-degree phase change that the rear driver radiation undergoes as it takes on its new reflex port output identity.

But there is one element of free lunch with reflex loading, and that's a less expensive bass driver. This is because, at the reflex resonance frequency, usually between 30Hz and 50Hz where there's typically lots of high-level programme material from bass guitars and kick drums, it's the port doing much of the acoustic heavy lifting rather than the driver (this is why port aerodynamics are so important). Diaphragm excursion demands on the driver are reduced significantly, and that potentially means money saved on the expensive components that make up the motor system: the magnet, pole-piece, top plate and back plate. So when Barefoot decided that the Footprint03 was to be compact and affordable, reflex loading must have been the obvious choice. As I've described many times in the past, though, there's reflex loading done well, and reflex loading done not quite so well.

Pass The Port

So which is it on the Footprint03? The signs are good. The port is tuned low, to around 37Hz by my measurements, which means its peak group delay will be below most important musical information. And the port itself is of large diameter, straight, and generously flared on both its entrance and exit. The volume of air that can flow in a port without turbulence directly influences how effectively it works, so generous diameter and flaring are vitally important. This is because when airflow becomes turbulent, a port, rather than contributing a significant portion of a monitor's low-frequency output, will lean increasingly towards behaving like a noisy leak. Compression, reduced bass output and potentially audible turbulence will result.

One result of the generous diameter of the port is that it needs to be relatively long, because the port tuning frequency substantially defines the low-frequency bandwidth limit of a reflex-loaded speaker; the longer the port, the lower the tuning frequency. In the Footprint03, therefore, the port reaches internally from its side-mounted position to not far short of the opposite inside wall of the cabinet. And because, below the port resonance frequency, a reflex-loaded speaker's output falls at a steep 24dB/octave, there's not even any practical opportunity to use equalisation to recover things, as to do so would require both huge levels of amplifier power and massive driver diaphragm excursion ability. Having said that, the Footprint03's published specifications describe an 18dB/octave low-frequency roll-off, so somewhat surprisingly, it seems the monitor EQ does indeed include a 6dB/octave boost below the port frequency. This potentially will eat significantly into the low-frequency amplifier headroom and driver excursion capability — although there's typically not much signal below 37Hz in most music.

So, the Footprint03 has a long port because it needs one. However, long ports can introduce a secondary, undesirable phenomenon, typically located well above the port's fundamental resonance, known as an organ-pipe resonance. I'll examine that later, but first, Diagram 1 illustrates the Footprint03's axial frequency response, measured at both 84dB and 94dB SPL at 1m (normalised so that they overlay). Measuring at two levels, one relatively quiet and one significantly louder, reveals if any level-dependent changes occur in the monitor (fast thermal compression, for example), and the results of the Footprint03 are impressive in this respect. The two curves of Diagram 1 overlay almost perfectly. Diagram 1 of course also reveals the axial frequency response of the Footprint03, and while it's not one of the very flattest, it looks perfectly acceptable, being within ±2dB from 100Hz to 15kHz. The slightly depressed region centred around 550Hz followed by the subtly raised presence band might result in a subjective signature sound, but the proof of that will be in the listening.

Moving on, Diagram 2 again shows the 94dB axial frequency response, this time accompanied by the second- and third-harmonic distortion curves. »

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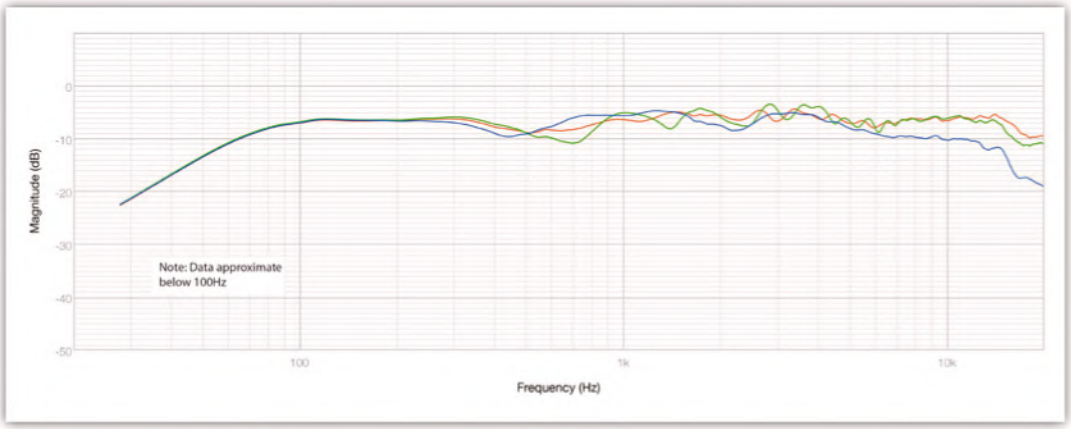

Diagram 3: Comparing the on-axis response (red trace) with 15-degree horizontally off-axis responses (HF and LF sides; green and blue traces, respectively).

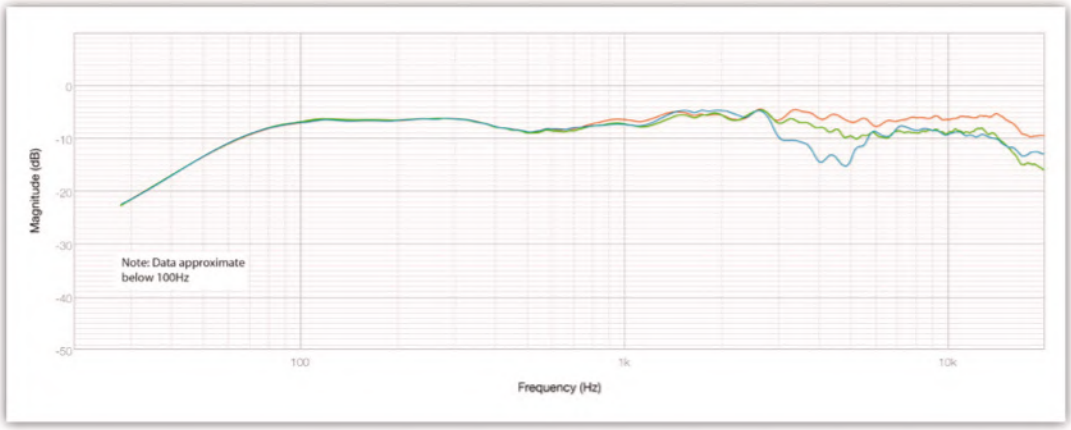

Diagram 4: As Diagram 3, but measured vertically off-axis, above (green) and below (blue).

» The Footprint03 again delivers an impressive result, with the more troublesome third-harmonic distortion remaining very low — typically 0.3 percent or below.

Diagrams 3 and 4 investigate how the Footprint03's frequency response changes 20 degrees off-axis. Diagram 3 illustrates the above- and below-axis curves (with the monitor in landscape orientation), and Diagram 4 illustrates the left (LF driver side) and right (MF/HF driver side) curves. Things to note are firstly that the relatively high bass-midrange crossover results in some off-axis variation down below 1kHz, secondly that the response is slightly tidier below axis than above, and thirdly that in most situations, where the monitors are installed in landscape orientation, it's probably advantageous to have the tweeter/midrange driver array located on the inside. Having said that, the generally impressive level of response consistency the Footprint03 shows off-axis means it potentially offers good installation versatility. For example, portrait orientation, with the midrange and HF driver array uppermost, might be a useful option in some situations and it would likely work perfectly well.

Finally, Diagram 5 reveals whether Barefoot have managed to keep the almost unavoidable organ-pipe resonance of long reflex ports under control, and the answer is that they have. A resonance is apparent at 550Hz but its peak is 15dB below the port output, so is unlikely to be specifically audible. For interest, I captured the port data at volume levels equivalent to 84dB and 94dB at 1m, and the overlay in Diagram 5 shows that either the port has begun to compress or that the Footprint03 has some bass driver protection limiting built into its electronics. The fact that the level loss mainly occurs below the port resonance suggests the latter — although low-frequency port noise was clearly audible when I made the 94dB measurement.

Listening In

So, the Footprint03 delivers an impressive set of measurements, it looks great, it's usefully compact and it's relatively affordable. But how does it sound? It sounds very good indeed.

Starting at the low end, the Footprint03 offers extended low-frequency bandwidth for its size, but also, to my ears, displays a hint of reflex character: a slight softness with a suggestion of overhang. But I think Barefoot have been careful not to be too greedy with the bandwidth extension that reflex loading can provide, and have kept a keen ear on its dynamic, pitch and time-domain characteristics, so the Footprint03's bass is eminently usable in a mix context. It offers easily adequate bandwidth extension for nearfield use but does so without adding serious distracting artefacts. The bass perhaps doesn't quite have the dynamic punch of the best closed-box monitors, but it's affordable and is undoubtedly capable of doing a perfectly competent monitoring job.

Moving up the frequency band, the midrange is notably clear and revealing, but tonally it feels mildly recessed at the lower end of the band. Voices in particular seem to lack a little warmth and body, making them sound slightly small and sit further back in the mix. A dB or so lift around 500Hz and a similar cut between 2kHz and 5kHz did the trick for me, and brought the balance more to where I like it.

ALTERNATIVES

High performance three-way monitors in the same price range as the Footprint03 are relatively rare, but the **ADAM A8H** and **Dynaudio LYD48** fall into that category and would be worth considering. If you're not tied to the three-way configuration however, two-way monitors in a similar price bracket such as the **Focal Solo 6**, **Neumann KH120 II** (or at a stretch the **KH150**) and **HEDD Type 07 Mk2** are worth a listen.

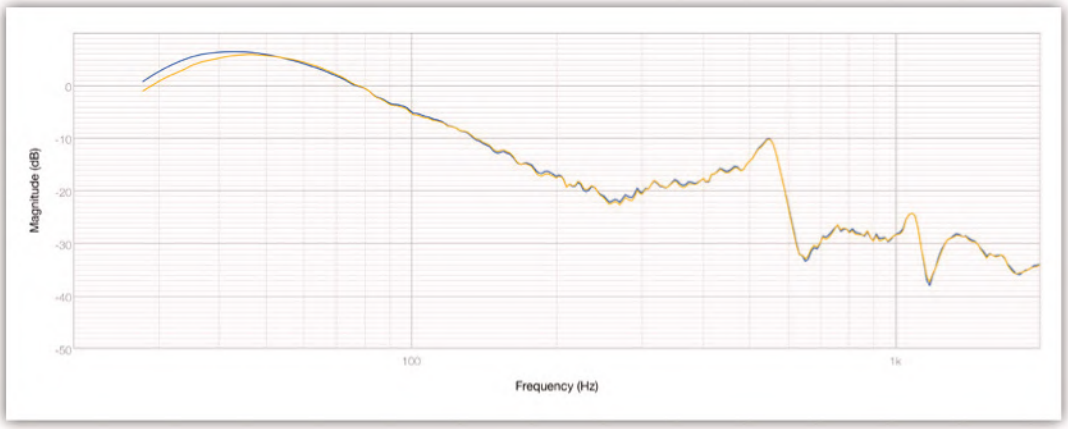

Diagram 5: Close-mic measurements of the port output, at 84dB SPL (blue) and 94dB SPL (yellow).

Your mileage may vary, however, and it's partly due to the Footprint being a fundamentally well engineered monitor that EQ can be applied to shape its sound. The Footprint03 isn't 'voiced' to disguise inherent flaws, as some speakers are, because it doesn't really have any such flaws.

Even without EQ, the Footprint03 midrange driver's very low distortion (probably a result of its high-tech motor system) is explicitly apparent. Its subjective performance is exceptionally clear and revealing, and it had me trawling through numerous old favourite tracks and mix sessions just to see what it made of them. For example, Athena Andreadis' beautifully recorded voice on her first album, *Athena*, was an absolute delight on the Footprint03. It explicitly resolved every last detail of the recording and mix, and did so with a wonderfully well-focused stereo image. The Footprint03 midrange has that rare ability to put everything in the right place and in the right proportion. This is invaluable in a mix context, I think. Another example of music that had me fixed to the spot was Laurie Anderson's *Strange Angels* album: the detail of her voice and how it's woven in and out of the mix was wonderfully revealed by the Footprint 03s.

Leaving On A High

Traditionally, having described bass and midrange, I'd move on to the Footprint03's high-frequency performance, but in this case there's not much to say other than that the tweeter performs in the manner that I've come to expect from the ring-radiator format. It just gets on with the job, offering great levels of detail and resolution, but without drawing attention to itself, and it sounds perfectly integrated with the rest of the Footprint03 system. And that word 'integrated' kind of sums up the whole Footprint03 experience for me. The Footprint03 looks, feels and sounds like a really well conceived and sorted monitor, one where every element is carefully considered and properly engineered.

Previous Barefoot monitors have all been significantly more expensive than the Footprint03, and expense, in some respects, tends to make the monitor design job less challenging. This is because expensive monitors are less constrained by engineering compromises than those of a competitive price. If you set out to make an inexpensive monitor, you'd better know exactly where to spend the manufacturing budget, and where you can make savings. My overriding impression of the Footprint03 is that, judged on those terms, it's an exceptionally capable and attractive monitor.

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Cong Burn Strokes

Software Instrument

At first glance the 2024-vintage Strokes looks like its MIDI-only predecessors. The four programmable note lanes and accent lane still dominate the interface, surrounded by pattern and note choosers, the Matrix and Weights modulation sources, Shares randomisers, and the colourful Scope, which visualises modulation values in real time. The multiple page choosers and Voice Modulation section at the bottom point to vastly expanded capabilities elsewhere though.

Strokes has evolved from a clever MIDI sequencer to a sophisticated and original instrument in it own right.

ROBIN BIGWOOD

When Strokes by Cong Burn (aka UK-based software developer, producer and label owner John Howes) was last reviewed in *SOS* just over a year ago it was a VST3 and Audio Unit plug-in for Windows and macOS, with an iOS-compatible AUv3 version on the horizon. It was also a sequencer, pure and simple, albeit an unusual one, with multiple drum-machine-like trigger lanes, algorithmic generative features, and step-driven modulation sources that ran inside your DAW, playing or modulating other virtual instruments. I thought it was brilliant, though you had to think laterally and know your DAW's MIDI routing capabilities inside out to get the most out of it. One trip around the sun later, and what a change there's been. So much so, that Strokes is easily worth another look.

Same Strokes/ Different Strokes

Strokes of 2024 (currently v13.3) is a development of what came before, so in fact there's still masses in common with earlier versions like the one I reviewed at www.soundonsound.com/reviews/cong-burn-strokes. That's still worth a read if you want to get up to speed on some of the fundamentals here, and especially the ins and outs of the unusual Matrix and Weights modulation sources and the Shares randomisers. Also, because all of what follows now is new, there's a lot of it, and some of it is pretty wild!

While the ability remains for Strokes to act as a MIDI sequencer and generator of MIDI CC messages, the headline news is that it can now run in a self-contained manner as well, generating its own sounds.

Crucially, all eight note channels can either play back one-shot or looping samples (mono or stereo, up to 32MB, in every audio format I tried including AAC and FLAC), or they can load one of the 24 synth engines of Mutable Instruments' classic, open-source (and sadly discontinued) Plaits Eurorack module. There's everything from virtual analogue tone generators, through FM, additive, wavetable and granular oscillator types, to physical modelling, speech synthesis, synthetic drums and a chiptune oscillator. All the synthesis models load instantly, and they're controlled with just four parameters that vary between models.

Common to sample and synth channels alike are a clutch of other sound-influencing parameters, including attack and release times for a simple attack-decay envelope, a multi-mode resonant filter, channel gain and pan position, and sends to the onboard reverb and delay. Those are fine by the way: in character the reverb is sparkly and spacious, the delay quite dark-sounding. Both can be usefully modulated, but you'll need to look to external alternatives in your DAW or iPad host to achieve more ambitious treatments, or even a ping-pong delay effect for example.

Standing Alone

One of the milestones in Strokes' recent development was its release as an app for the iPad, and as an AUv3 that works beautifully in a host like AUM. There it'll act as a sound generator or can trigger other AUv3s or external gear from its MIDI outs. Or both.

On Windows and macOS Strokes is a VST3 or Audio Unit plug-in only, and it relies on a DAW host for basic transport controls. But then, as on iPadOS, it'll similarly function as a MIDI source and sound generator.

In any OS, Strokes offers individual audio outs from its channels in addition to a stereo mix, so it's easy to add effects beyond the onboard reverb and delay. Its projects are cross-platform too, so you can start work on an iPad and finish on a Mac, for example. Sadly project files don't encapsulate sample data, but you can work around this seamlessly by storing samples on cloud services, like Apple's iCloud drive.

The synth engines are just the start, though. Each channel/synth can also now accept up to nine simultaneous modulation streams, controlling various basic parameters like pitch (which is always constrained to scale notes, user-selectable, to keep things musically coherent), envelope decay, and send levels. Additionally you can modulate specific parameters for each synth engine (clearly named in the interface, rather than being labelled generically, thank goodness), and for sampler channels modulate the sample start and end points, and bit-depth and sample-rate reduction. Modulation sources are the Weights and Matrix sources I described above, the accent channel, the channel's envelope generator, and three preset mixes of the envelope and keyboard tracking.

Stroking, Not Drowning

Essentially, where Strokes of old already had an ability to generate interesting material that tended to be different from what you might generate conventionally, working in a more linear fashion in a DAW, the new features magnify that hugely.

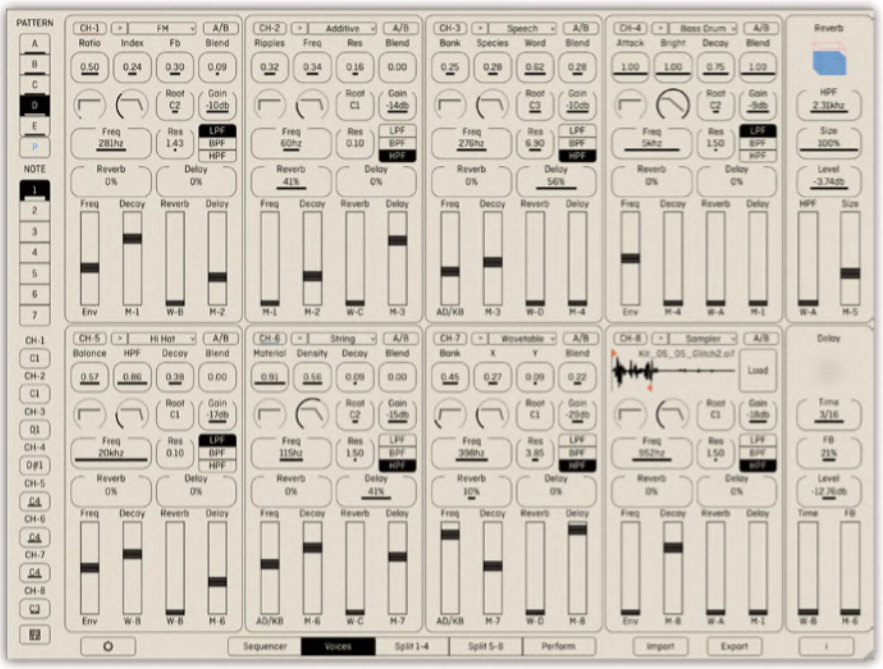
The onboard sound generation is key in this: it's just so much quicker, more immediate and convenient to work 'in the box' than have to configure virtual instruments in your DAW or iOS host, and it helps that the Plaits synth models frequently sound great, often with a sort of raw, contemporary character.

There's also something unique about Strokes' modulation scheme. Unlike conventional grooveboxes that might utilise LFOs, programmable pattern sequencers and recordable knob movements (and so on) to enliven repetitive pattern data, Strokes has nothing this traditional or predictable: modulation streams are always fundamentally interrelated to note data, and consequently notes and modulation sources interact in an organismic way. It can really feel alive, and it will certainly challenge most users to think differently.

Take for example a typical ducking effect you might set up in a DAW, with a bass drum keying the side-chain of a compressor applied to a synth sound: Strokes can do that (via a Weights channel, triggered by a bass drum channel, modulating the synth channel's gain), and masses more besides, in dozens of directions at once if you wish. It can lead to creations that sound desirably contemporary, and I personally found it hugely addictive: the system feels constantly prone to happy accidents, but is still very controllable. Now, much more than before, Strokes feels like a complete sequencing environment, self-sufficient, synth-oriented, that you might be quite content to work within either entirely, or until later stages of a production that eventually needs some vocals or live instruments, and other more deterministic input.

Performance Enhancement

The last groundbreaking feature in Strokes v3.13 is the Perform view. This visually busy page with its trio of X/Y pads and bank of crossfaders is built for live »



The Voices page, seen here in one of the new alternative colour themes, is Strokes' onboard sound-generation engine room. Each of the eight note channels can generate MIDI, replay samples, or become one of the 24 synth models of Mutable Instruments' classic Plaits module. All channels have an identical, simple but flexible synthesis architecture, and accept multiple modulation streams.

Cong Burn Strokes
$60

PROS

- Now a self-sufficient production environment, with synths, samplers and effects.
- Perform view real-time tweaking facilities are weird and wonderful.
- Remains lean and responsive: new features haven't clouded the original concept.

CONS

- With no hold stage, synth/sampler envelopes are essentially percussive in nature...
- ...as are sequencer steps, which can't be tied, and carry no duration, velocity or note data.
- Note quantisers let Strokes enter melodic and harmonic realms, but the per-channel implementation can feel ponderous, and they can't be turned off.

SUMMARY

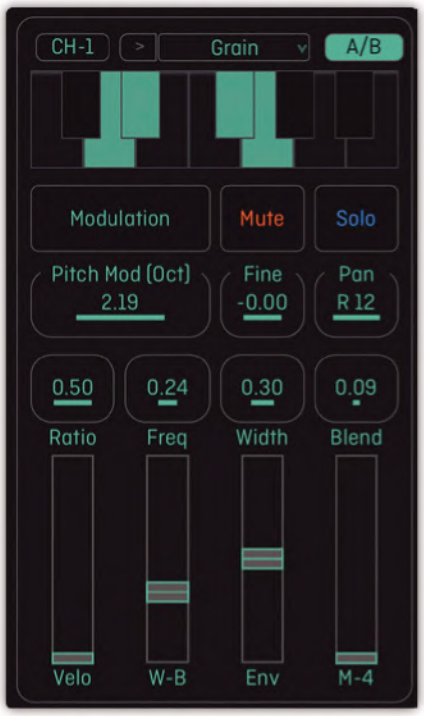
A beautifully conceived app and plug-in that's more inspiring than ever. Strokes now supplements potent step and Euclidean sequencing with sample replay and the sounds of one of the Eurorack world's most useful and well-loved multi-engine synths.

» tweaking. As with other aspects of Strokes, though, all is not quite what it seems.

Easiest to grasp are some simple mirrors of groups of parameters that would otherwise be strewn across other pages: channel enables/mutes, the global reverb and delay, the Shares randomisers, the Matrix and its value knobs, channel note selection, and basic sampler controls. Also there's a master saturator and compressor here — Heat and Glue — and they will remain active even if you navigate away from this page and do all your work elsewhere. That's a particularly nice touch for anyone working standalone, to allow for some hotter and more controlled master output levels.

But those crossfaders: they don't fade audio signals, but interpolate parameter values and note triggers between two patterns. One role for them is to let you morph smoothly and slowly between even radically dissimilar patterns, that could have quite different settings and modulation schemes. Or you could 'play' the crossfaders, to morph to different pattern content sporadically, rhythmically, on backbeats for example: the system is very responsive. What it can't do, incidentally, is smoothly crossfade between dissimilar synth engines, or parameters with only on/off states, or discrete modes (like the filters). John Howes told me a scheme that could do this was once considered, but that it was too CPU-intensive, requiring multiple instances of each channel's synth engine to be running concurrently.

However, there's also creative potential in the crossfaders, because at points along the spectrum between two existing patterns can exist great, new, interpolated material that you didn't quite generate yourself, but that you seeded and Strokes cultivated, so to speak. As you explore like this, the dozens of changing parameter values in the background are represented above the bank of crossfaders by a little visual squiggle (somehow reminiscent of images of DNA and viruses) in a box. And those boxes are draggable: grab your freshly-mutated pattern with mouse or finger, drop it to one of the five pattern slots to the left or right, and it becomes a new, legitimate, recallable pattern that itself can spawn new patterns.



An alternative 'B' view of each voice reveals a note quantiser and additional modulation options.

What's great about this system is that it lets you go on creative journeys even if you only have a couple of patterns at starting points. Also, that it makes pre-programmed sequences that might otherwise feel rigid actually very fluid. There's no way to record what you do (other than literally record the various audio outputs from Strokes, as you tweak) but that's very much the point of it: it's a real-time, live performance feature, not something to be programmed.

Digging in a little further, there's some real subtlety on offer. First off, the uppermost crossfader has a 'master' role, in that it adjusts all 12 beneath it simultaneously. But each can be operated individually, to fade just a single voice channel or interpolate modulation and effects settings. Also, while it could seem that Perform page work obliterates your carefully-honed patterns — after all, five pattern slots is not many — in fact it doesn't. Existing patterns are loaded initially, but thereafter what happens in the Perform page stays in the Perform page: go back to the Sequencer view and you'll now find a flashing 'P' under your pattern slots, showing that Perform page temporary operations currently have precedence. When you again select patterns A-E you'll find them exactly as you left them.

As for the two square X/Y pads, these offer functionality that is at least a little more familiar. On them you'll find numbered, draggable 'pucks', relating to the programmable channels 1-4, logic channels 5-8, or all eight at once. What the positions of these pucks denote are the current channel modulation amounts (X axis) and the static values (Y axis) for the parameter chosen in the pop-up menu above. By dragging the pucks directly, you can very much 'play' parameters in real time.

That just leaves the central, octagonal pad. This is a sort of relative value adjuster, with an intriguing geometric bent. Again, you choose the channel parameter it'll control, on all eight note channels simultaneously, in the pop-up menu. Then, as you drag the puck away from its neutral middle position, cobweb lines appear representing parameter value changes. Channel numbers you move towards have their parameter values increased, and the ones you leave behind have them decreased. It's impossible to generalise what can be achieved here, because it's so dependent on exactly what you have loaded to each channel, and your starting point parameter values. Working with filter cutoff frequency, for example (and assuming you're running low-pass filters on each channel), it could brighten the channels you move towards, spotlighting them and making them louder, while darkening others. Just like the X/Y pads, it's very responsive, supporting flick-type gestures to make fast-moving changes in sound if necessary, and working in either a momentary or latching way.

Meta-crossfading

Here's another remarkable, mind-bending feature: crossfades to totally different projects! The idea is that while you work on the patterns of your current project in the Perform view, you can load a different project and access its five patterns and seven note patterns alongside. The current project remains in memory and keeps playing. But the new project content can be dragged into any of the 'current' slots, to become potential targets for a crossfade.

An example of this in use would be running a whole live set of several songs/projects, without ever stopping playback. As one project continues you can gradually cue up the content of another (and eventually another, and another) and sashay your way between them seamlessly.

Master Strokes

As I mentioned in last year's review, Strokes occupies a pretty much unique position amongst software tools, with an environment, workflow, and underlying feel that's much nearer to a well-stocked modular/Eurorack case than to most DAWs. Consequently, it's not what you'll naturally choose if you're into acoustic »

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» styles or rock music: it's unashamedly electronic, specialised towards the minimal, experimental and glitchy.

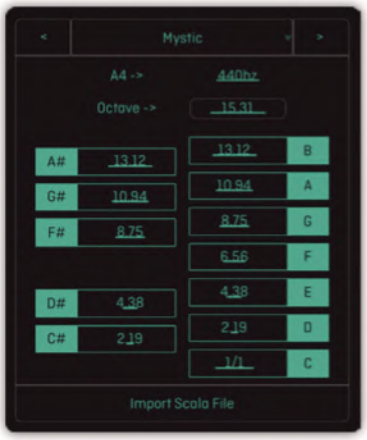
It's also full of apparent limitations. Note lane triggers are strictly monophonic and purely percussive, with no concept of note pitch or velocity of their own, and no possibility to tie grid steps to achieve notes of longer duration, or program ratchets. The only way to achieve changes in pitch is through loading of pre-defined note sets, modulation, or in a more limited and very specific way via live input from a MIDI controller. This aspect alone can pull you up sharply, to begin with at least, when first transitioning from a traditional DAW.

I could go further. The maximum of only five patterns per project really focuses the mind, as do the synths' and sampler's rather limiting AD envelope with no hold stage and brief (circa 3 seconds) maximum release times that preclude generating any sustaining atmospheres or 'pads'. I'm sorry the channel pitch quantisers can't be disabled: although the new tuning features can introduce microtonal spice you can never break away from the concept of scales completely (as you easily can in Eurorack, for example). Conversely, just duplicating a note set with the intent of quickly transposing pitches to explore other harmonies is a real faff if you have note quantisers in play, requiring you to go through and reconfigure them on every single channel. All of this contrives to make Strokes a better playground for rhythms than for melodies or harmonies.

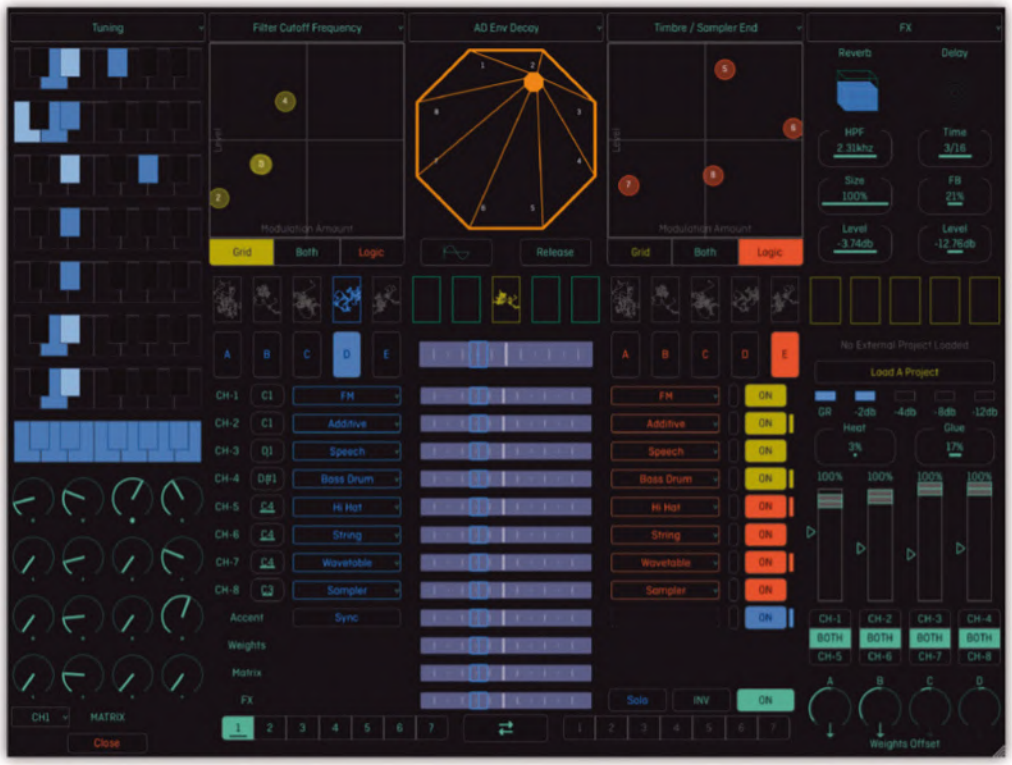
However, the idea is that limitation stimulates creativity: that force burns bright in Strokes. The particularity of the design and undoubted degree of quirkiness here channels users into ways of working that are very different to traditional, linear-leaning DAWs, and can be simultaneously challenging and exhilarating. I've mentioned the similarity in character to Eurorack modular elsewhere, and that observation is more valid now than ever in Strokes, with the multiple Plaits engines and modulation paths running all over. You would need to invest heavily to create a rack that could replicate even a small part of what Strokes can achieve, and it would struggle to ever match its ease of integration with DAWs and MIDI sound sources, both virtual and tangible. I wonder, in fact, if for many users Strokes might actually prove the more enjoyable and fruitful option.

All told, Strokes of 2024 is a huge success. Compared to some plug-ins it's downright cheap, but it's also one of the most interesting electronic music playgrounds I can think of. Despite a few interface concepts that may not be immediately intuitive it's surprisingly quick to learn and get nice results from: a good, succinct PDF manual helps with that, as does a great overview video on the Cong Burn website. Working with Strokes, I frequently got the impression that it was giving back at least as much as I put in, which is a rewarding and energising thing to experience, and actually the mark of any top-class musical instrument. With the new sample and synth engines, and Perform page, a vast amount of musical potential has been added without compromising immediacy, and with a nice sense of balance between what can be now be achieved 'in the box' and what's possible when hooking up to a wider ecosystem. There's creative potential, open-endedness and aliveness here that few other single plug-ins and iPad apps can match. ■■■

Scales



To complement its many unconventional onboard sound sources, Strokes can now map its internal note triggers to alternative scales and tunings described in Scala format. Ten are built in, and there's enough flexibility to constrain the pitch palette to pentatonic or modal scales, to tune individual semitones, or describe systems (as shown here) based on non-consonant octaves.



The colourful new Perform page is a tweaker's delight, giving direct real-time control over crucial parameters, for all channels at once. The crossfaders interpolate parameter values between two chosen patterns, and can either be 'played' or mined for new starting points.

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MODULAR

Expert Sleepers Pandora, Cicely, Amelia, Aloysius & Otterly

Eurorack Modules

It's not often we have to issue corrections here at *SOS*, but I need to make a confession. When I reviewed Expert Sleepers' initial slew of all-analogue modules not long ago, I was all too happy to make quips about their names — Ivo, Lorelei, Beatrix and Persephone — sounding like they belong in a children's book, not realising at the time that those names were in fact Andrew 'Os' Ostler's homage to the Cocteau Twins' classic 1984 album *Treasure*. With that gap in my music knowledge very much exposed and since filled, we can move on to the next batch of analogue Expert Sleepers modules, which expands the line-up in several new and exciting directions, while also not being afraid to forge deeper into some areas already explored elsewhere in the range. In this sense, a fully self-sufficient system is clearly on the cards, but doesn't seem to be top of the agenda.

First up is Pandora; a "band-pass filter and distortion processor", though really it does far more than that. It crackles with character and attitude (befitting of its name's extra-Cocteau associations), boasting filtering, distortion, VCA-like behaviour and even a mode to moonlight as a percussion module. Its vactrol-based Alpha and Beta knobs ostensibly control low-pass and high-pass filtering, though both are in fact moving the centre frequency of the broad-curve band-pass filter to create something quite different. There's also gain, feedback and a wet/dry mix — all with their own CV inputs. If there was one module in the series that could be described as multifunctional, I'd say this is it.

Of course, despite being a band-pass filter with both Alpha and Beta used together, it can perform well in either a low-pass or high-pass capacity; though possibly a little more idiosyncratically than you might want if it's the only filter in your case. It certainly calls to mind synths like the Korg MS-20, whose drivable low- and high-pass filters account for the lion's share of its growling, throaty character, something Pandora has in spades. Its gain circuit is driven by a pair of beastly MOSFET transistors, and at gentler settings (with Mix set to full) can be used like a modestly saturated VCA, while at extreme settings can crank things into aggressive distortion.

On top of all this, the Pandora also has a Low/High switch. This significantly lowers the filter's centre frequency range and in effect turns it into a capably gnarly drum synthesizer. Of course, this is theoretically possible with any filter; but here it's presented on a platter, and together with the onboard distortion amounts to something of a masterstroke. Send triggers into the input to ping the filter and adjust the feedback, distortion and filter frequency to create a range of percussion sounds that are smooth, crunchy and everything in between. Beautiful.

Things get really gnarly on the distortion front with Cicely, an "octave fuzz" based on Roger Mayer's famously Hendrix-favoured guitar pedal, the Octavia. While the idea of an octave fuzz sounds like a simple composite effect, in reality the pedal did — and this module does — something much more interesting. A simple wave or pure tone will appear to jump an octave with the effect at fully wet — far from alien for those used to playing with distortion, but more complex audio will take on a more complex kind of metallic sheen, offer strange harmonics and then some. After experimenting with a few sounds and sequences from within my system, I fed my Rickenbacker 360 into the Cicely and found it predictably gritty, weighty and generally excellent-sounding.

The Cicely has a few added functions to make it worth the trip to Eurorack. Aside from the obvious CV control, it has a handy envelope follower with both positive and inverted responses; particularly useful when putting things like a guitar through the Cicely, since it creates a CV equivalent of your playing to patch elsewhere. There's a useful Centre control to create asymmetrical clipping, which amounts to something like pulse-width modulation, and there's an extra audio input for summing — cramming, I should say — even more harmonics through the circuit. With the Mix control, this can be used to add subtle colour or, if you'd rather, destroy your sounds completely. It's worth pointing out that the Cicely makes a very good partner for the Pandora, whose unique filter circuit was perfect for smoothing off occasional top-end harshness or taking some mud out of the low end, or both. I didn't even hate the result of driving both gain circuits at the same time.

Aloysius and Amelia are a brace of envelope generators offering different takes on a four-stage envelope design. Suffice it to say: while it may be more familiar and feel the most instinctive, more traditional ADSR envelopes are really best suited for manual playing, while variations on this (that might be better suited to sequencing or cycling) usually end up being simplified to ADR or AR envelopes.

Almost identical in their layout and aesthetic, both Aloysius and Amelia can be set to cycle like a complex LFO, both have controls for particular stages' response curves, and both offer different modes to dictate how — or if — their respective stages respond to incoming gates. It's also possible to switch between slow, medium and fast time ranges for different stages.

So how do they differ? In a nutshell, Aloysius is an AHDW envelope, while Amelia is an ADBR envelope. Don't worry, I'll explain. AHDW stands for attack-hold-decay-wait, which is also known as a trapezoid generator and closely associated with the EMS synths of legend. It's worth noting that all of these are time values, unlike ADSR where sustain is a level value. Upon receiving a trigger, its attack stage rises to maximum according to the attack time and curve, where it is held for the duration of the hold time. It then decays, according to the decay time and curve, until it reaches 0V again and then waits according to the wait time before retriggering. It's particularly good for classy, sparse, cycling envelopes with its wait stage, and also great for generating movement with a variety of gate lengths, or conversely for using with triggers of any length in more of a slew-limiter-type capacity.

Amelia, on the other hand, is an attack-decay-break-Release envelope. The key here is that break represents a level value; namely that of the decay stage, at which point the envelope moves onto the release stage. I found it helpful to think of break almost as a mix control between the decay and release stages. A high break value will mean that the decay stage will only occupy a fraction of the envelope response, and *vice versa*; meaning that at one pole it essentially renders the Amelia an AD envelope. At the other, AR. It's very well-suited to creating Buchla-esque, plucky but natural-sounding responses to simple triggers, so is likely to waste no time making best friends with your sequencer.

Lastly, and most recently unveiled, is the Otterley. A multi-LFO module, it offers an impressive five different LFOs along with the ability to 'spread' their frequencies to become more and more asynchronous. The first four of these have their speed, polarity and waveform controlled together, while the fifth operates independently of these with its own speed knob. It's a simple and immensely useful module, switchable between square and sine waves (though these are customisable via a few trim pots on the PCB) and with a wide range of modulation through positive or negative values, or both. It's also visually quite lovely with its undulating red and blue jacks. I found it incredibly rewarding feeding all of its LFOs to the various CV inputs of one synth voice; instantly creating a holistic, natural and musical-sounding ebb and flow.

Having finally made the Cocteau Twins connection, it immediately occurred to me that Expert Sleepers have stopped short of the final track on the album, 'Donimo'. Perhaps there's something special up Os's sleeve to conclude the run? Though since he has also made recent mention of further album-themed series, I also wouldn't be surprised if this *Treasure* range is understatedly rounded off with another sleek circuit in elegant 8HP. In any case, if the first batch of analogue modules didn't demonstrate that by now Andrew Ostler can more or less do whatever he wants, this second one has. There's nary a dud here, and with these modules' prices averaging around $199 — not far off pedal-price — I'll offer they're something of a steal.

William Stokes

$ From $185.

W www.expert-sleepers.co.uk

What's New

Spanish modularists **Befaco** hosted a DIY workshop in London at the end of April, hosted by **The Synth Shed** and with support from **Thonk**. Befaco brought over 50 kits with them for guests to build; from the dual-channel wavefolder Chopping Kinky to the three-channel noise generator Noise Plethora. Be sure to keep up to date on future events via the Befaco and The Synth Shed websites: www.befaco.org www.thesynthshed.com

Speaking of noise — and in more ways than one — **Noise Engineering** have unveiled yet another powerful firmware for owners of their Legio module platform: a noise generator. Morbus Legio is a rather interesting combination of both processor and noise generator; capable of both processing external audio with clipping, wavefolding, distortion and chorus, as well as noise that can be spread across the stereo field via CV. www.noiseengineering.us

Clank have announced the Endline, an upgraded version of their preceding SUM module, which boasts four stereo channels of summing, submixing, two sends and returns, a headphone output and transformer-balanced outputs — all in 10HP. www.clank.eu

Erica Synths have also been at work in the signal processing department: the latest in their range of redesigned Hexinverter modules is the Mutant Hot Glue, a four-channel bus mixer designed for analogue effects chain design. It features a compressor with series or parallel compression and a side-chain input, as well voltage-controlled "acid-style" distortion. www.ericasynths.lv

William Stokes

» Modular Interview

Expert Sleepers' Andrew Ostler gives us a blow-by-blow account of the thinking behind his new analogue modules.

Photo: Viktor Shkarev

On Pandora

Pandora's a bit of an odd one. The germ of the idea was that of using a CMOS inverter IC as a high-gain analogue amplifier, as in the classic EDP Wasp's filter, but here I've done it with discrete power MOSFET transistors instead of the logic chip. These form a band-pass filter, much like that in an 808 kick-drum circuit, which is why Pandora has that switch to drop the frequency range. And then of course it has vactrols for extra flavour. I've made some wild rhythm tracks by feeding Pandora noise and a load of rhythmic CV modulation.

On Cicely

The Octavia pedal is a particular favourite of my long-standing collaborator Mike — we have a band called Darkroom. As with many of these vintage '60s pedal designs, it's fundamentally a really simple circuit, but of course not voltage-controllable. So the challenge here was to make it CV-able and therefore useful in a modular context. I love how this module adds grit to the sound. So many great electronic timbres are low-pass filtered — and who doesn't love a classic VCF? — but that can leave them lacking in top end.

On Aloysius & Amelia

In a wonderful bit of serendipity, there are two adjacent 'A's in the naming scheme for these module. Aloysius took the core of Amelia and became something inspired by the EMS Synthi's trapezoid generator. It's an extremely flexible source of modulation, especially in 'free-running' mode when controlling all the various stages with CV.

Amelia was designed to be particularly useful in a sequencer-driven patch, where you often have trigger pulses flying about but don't have or don't want to worry about an actual gate to define the note length. It generates its full envelope shape from a trigger, even if the attack is quite slow. It excels particularly at 'plucky' sounds, where you have a fast initial decay and then a longer 'ring,' not unlike that of striking a low-pass gate — the 'Buchla Bongo'!

On Otterley

It was becoming apparent that this analogue range had almost become a complete 'system' by itself, which was never really the intention. The one thing lacking in such a system was LFOs, so the Otterley satisfies that need. The idea of 'spread' is one I've used many times in the past, going right back to my VST plug-in products. It's a really convenient way of controlling a number of things at once with one parameter. There's a lot of fun to be had with Otterley by patching it for self-modulation. *William Stokes*

Guest Technique: Total Dist-Os-tion!

It turns out I have a bit of a fondness for distortion — Cicely is obviously entirely that, but Pandora will happily crush sounds too, and Persephone offers some nice analogue saturation. One effect I'm particularly keen on is that of putting a short delay, maybe 300ms, before a distortion, and then playing some fairly long notes into it. You get some wonderful sounds when changing from one note to another, especially if they're quite close. The distortion obviously does its thing to the sustained notes as well, but when the notes briefly overlap because of the delay, the non-linearity of the distortion interacts with the non-aligned partials of the two notes to give lovely intermodulation 'wobbles'. It's a fascinating sound, and better yet, because it's only when the notes change, it's not a constant overwhelming noise, just a bit of sonic 'spice'.

Another favourite trick at the moment is to use my 'stereo' mixer as a dual-mono mixer — one channel is clean, while the other runs through a distortion. Then the pan pots become blend controls between fully clean and fully distorted, which can be applied to everything running through the mixer. Use with care! *Andrew Ostler*

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JBL IRX ONE

Active Column PA

It might be the baby of JBL's mini line-array range, but its small footprint and easy portability could mean the IRX ONE is just what's needed for smaller shows.

MIKE CROFTS

JBL have long included 'mini array' column speakers in their product range. The current line-up has a full-featured, all-dancing model called the PRX ONE, the more affordable EON ONE PRO, and now the IRX ONE, which extends the range by offering the same form factor and JBL engineering in a more compact and very portable package, with fewer bells and whistles. To call the IRX ONE a starter or entry-level unit is probably not wrong, but at the same time, for many applications it's also the optimum choice if you don't need the extra features and output level of the more expensive models.

Features & Specifications

The IRX ONE consists of a base unit containing an eight-inch reflex-loaded subwoofer, 1300W Class D amplifier module, and three-channel mixer (four if you count the Bluetooth input). Mounted above the base unit is a height-adjustable column, which incorporates six two-inch drivers. The system is capable of delivering maximum SPL of 118dB, and the whole thing packs into the base unit for one-hand carrying, weighing only around 15kg.

The control panel gives access to all the functions apart from the IEC power connector, and is handily located under a flap that also serves as the carrying handle when clicked into the down position. When raised, the cover/handle rests against the speaker column and snicks into place by means of a concealed magnetic latch — this cover can then also be used as a very neat holder for a media player such as an iPad or similar. The three mixer channels are analogue, and provide the basic controls necessary: input trim, two-band EQ and channel level to the mix. Phantom power can be applied to channels 1 and 2 and is switched globally. Channel 3 has a line/instrument jack input, and the Bluetooth input has no surface controls other than the switching/pairing button. The DSP pack for the IRX ONE includes automatic feedback suppression (AFS) and auto-ducking, whereby mic channels duck the Bluetooth channel. As well as the simple two-band EQ on each analogue channel, there's an overall rotary switch that offers five tone-shaping presets, and lastly there's a balanced mix output for daisy-chaining, AV feed or whatever expansion might be needed.

First Impressions

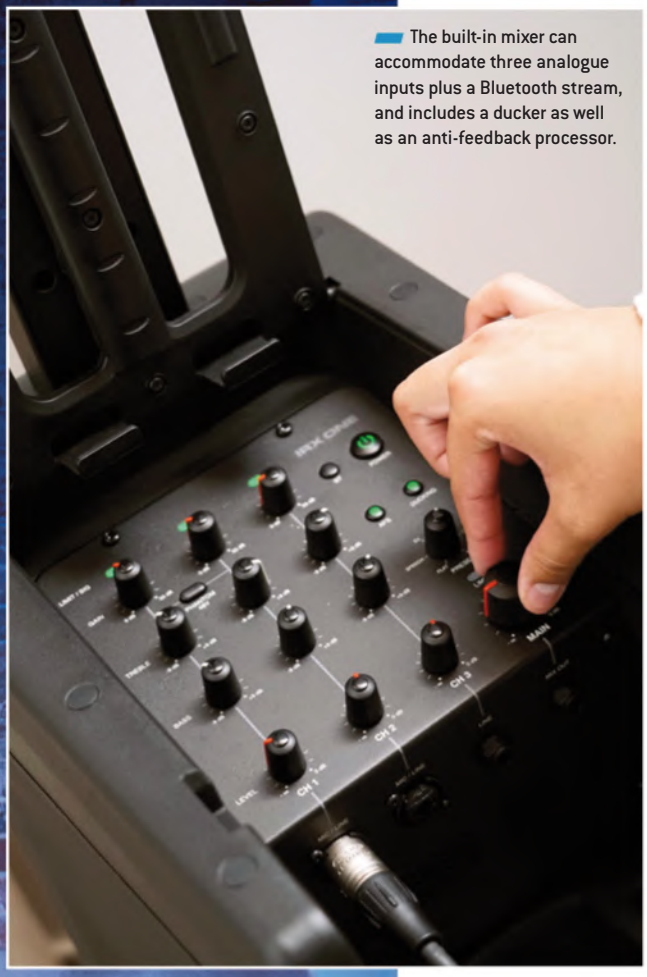
The IRX ONE is ridiculously easy to set up, and has been designed to be completely foolproof in use. The three parts of the column will only fit together when offered up correctly — you can't hook them up the wrong way round — and there are no cables involved other than the power lead. The carrying handle simply flips up to reveal the mixer panel and the three 'stick' sections are neatly stored inside. Only one of the three sections contains the actual drivers, and this can be mounted directly on to the subwoofer unit or you

JBL IRX ONE
$799

PROS
- Easy to carry, set up and use; it's almost impossible to get it wrong.
- Smooth, clear sound with excellent wide coverage.
- Easy Bluetooth hook-up, and the ducker and AFS functions are simple to use.

CONS
- No onboard reverb.

SUMMARY
The IRX ONE is provides quality sound with great coverage in an exceptionally easy-to-use, highly portable package. I'd imagine that it will find much favour in the educational and corporate worlds as it is an ideal AV tool, equally at home in a presentation, exhibition or public address context as it is with live music.



The built-in mixer can accommodate three analogue inputs plus a Bluetooth stream, and includes a ducker as well as an anti-feedback processor.

can use one or both of the other dummy sections as spacers to obtain more height. The built-in electronic cleverness senses how many (if any) extender sections have been connected and makes a subtle adjustment to the sound output to ensure optimum balance and coverage. The electrical hook-up between the three parts is achieved using four-pin recessed connectors that carry the output to the speakers; I measured a few Ohms resistance input-to-output on one of the pins in each section, and this presumably has some bearing on how the IRX ONE senses how many spacers are in use. As with just about everything else on the IRX ONE there's nothing for the user to worry about: everything is taken care of!

Having assembled the pieces and powered up the IRX ONE, I paired my iPhone so I could use the Bluetooth channel. Pairing is quick and easy, but you do need to bear in mind that the Bluetooth channel doesn't have a physical level control on the mixer panel — input level is controlled directly from the paired device, so it's as well to make sure that the master level is turned low to avoid sudden loud output. I mention this from experience, and will be more careful next time.

The overall output level is quite enough for a small event (AV presentation, house party, rehearsal, what have you) and the IRX ONE is forgiving of very hot input signals. Even at absolute full power it still sounds clean and clear, as if there's headroom left in the system, and the source device is likely to start working too hard before the IRX ONE does.

The mixer panel is simple and clear — what you see is what you get, and there are no hidden menus or multiple-function »

» controls to confuse an inexperienced or non-technical user. If LCD screens aren't your sort of thing, you'll be right at home with the simplicity of the IRX ONE. The only thing to be aware of is the way that phantom power is switched. Unlike the AFS button or the ducking button, which just click on and off when pressed, the phantom power has to be switched in a particular way, which at the time of writing isn't spelled out in the user guide. To prevent accidental switching either on or off, there's a short delay after you press the button — if you do a short press nothing happens, and if you hold the button for too long nothing happens. The button has to be held in for between one and two seconds. This did catch me out initially, and isn't specifically described in the user guide. It's actually a smart feature and quite deliberate: if you press the phantom power button and then realise that you really didn't mean to apply 48 Volts after all (hands up who's never ever done that?), you can just keep it pressed in for a couple more seconds and the operation is safely cancelled! Whilst on the subject of button pressing, the Bluetooth button also needs to be held for a couple of seconds to initiate pairing.

The 'bonus features' included with the IRX ONE are a ducker circuit, which applies automatic gain reduction to the Bluetooth music signal when the mic inputs exceed a certain threshold level, and an automatic feedback suppression circuit that does exactly that: detects audio feedback and quickly kills it by introducing a narrow notch filter, so you can have a microphone right in front of the speakers without worrying about loud squealing noises. In practical terms this is a great fail-safe feature, although the nature of these mini-array speakers makes unwanted feedback much less of a problem than with conventional full-range speaker setups. The IRX ONE doesn't include an onboard reverb effect, which is a shame for anyone intending to use it mainly as a portable vocal rig; if there'd been a pick-your-own feature list, I think I'd have chosen a basic reverb over the AFS function, but that's only based on what I personally would tend to use it for.

In Use

I took the IRX ONE along to a couple of school concerts, where a small footprint and wide coverage were very high on the wish list. The first event had only piano accompaniment, with various children performing acoustically across the front of a large-ish hall; there was no need to use any mics so we just used the IRX ONE as a keyboard amplifier, where it provided lovely room-filling coverage without sounding too loud in any one spot. The very wide dispersion and even sound balance was just what we needed, as the young performers were relying on being able to hear the accompaniment without it drowning out their voices.

"It provided lovely room-filling coverage without sounding too loud in any one spot. The very wide dispersion and even sound balance was just what we needed..."

The second school concert involved a similar hall setup, but this time I fed a couple of ambient capacitor mics into the XLR inputs and used the Bluetooth channel for pre-recorded backing tracks. Having only a single IRX ONE unit, I placed it at one of the rear corners of the low stage, and used both spacers to get the speakers above the performers' heads. As before, the sound coverage was very good and even throughout the hall, with performers and audience able to hear everything they needed. The ease of setup and smooth, clear output particularly impressed the staff; by the time I packed up and left they were asking about the cost of the IRX ONE, and discussing its suitability as a potential replacement for their ancient 'music trolley'.

The whole system can be packed up into the base unit.

I also took the IRX ONE along to a couple of band rehearsals, where I used it as a vocal monitor in place of our usual EON ONE Compact. The main difference was that the Compact is used as a floor monitor and most of the (large, 14-piece) band can't hear it very well, whereas the IRX ONE facing backwards straight at the band let everyone hear the vocals due to the super-wide dispersion of the speaker array. It wouldn't have been powerful enough for a live gig with this line-up, but for rehearsal, even with the band giving it fairly large, it was great as a monitor: crisp and clear and no hint of feedback, even without the AFS deployed. I found the two-band EQ perfectly adequate for all I tried, and I did experiment with the EQ tuning presets. These do pretty much what you'd expect from the descriptions; as well as the usual Flat and Speech settings, there are options called DJ, Café, Bright and KTV (I'm assuming that KTV stands for Karaoke TV?): all useful options, although I always ended up back at Flat.

My final live test was supplying something for a 'corner stage' gig in a café bar, for live solo vocal and backing track. This is, in live music terms, exactly what systems such as the IRX ONE are put on the planet to do. I didn't need paint-peeling output level, I only needed a couple of inputs, and the setup had to look unobtrusive and take up almost no floor space. The IRX ONE was the perfect fit in all respects (although I did need to hook up an external mini-mixer as an external reverb for the vocal) and the same trademark clarity and extra-wide sweet spot were exactly what the room required. The fact that I carried it up a flight of stairs in one hand, with a mic stand in the other and a small carry bag for 'bits' on my shoulder, further enhanced the overall experience!

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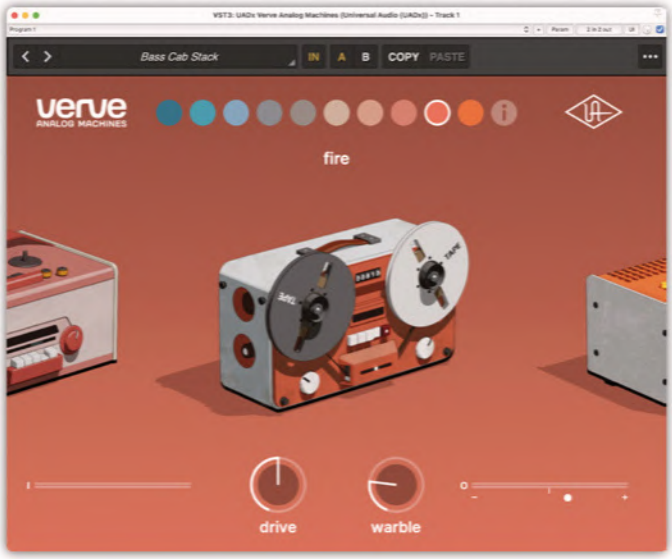
Universal Audio Verve Analog Machines

Saturation Plug-ins

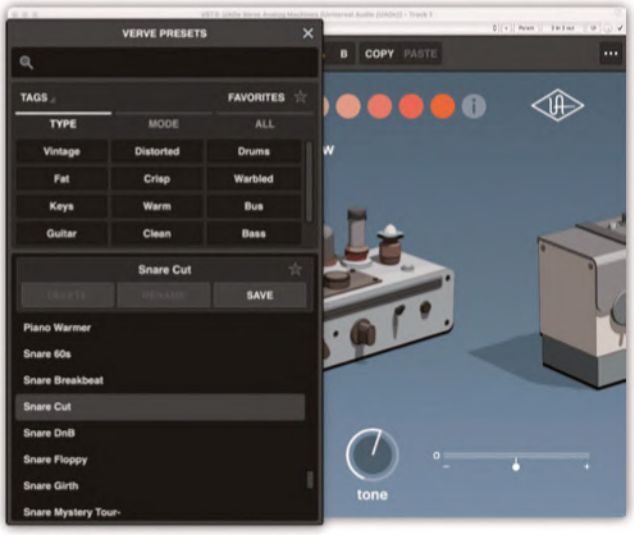
UA have an enviable track record of creating analogue modelling plug-ins. But typically these have been emulations of specific gear that offer the user lots of control. Verve Analog Machines and its 'lite' sibling Verve Analog Machines Essentials tread a different path: they present a choice of different emulated devices, each of which has only one or two controls that allow you to refine the sound. Both versions are available for desktop Mac and Windows hosts and run natively (you can't run them on a UA DSP interface or accelerator card).

These plug-ins are all about warmth and distortion, and the flavours on offer range from gentle saturation at the subtler end of the scale through to crunching overload and squelchy obliteration. In Verve Analog Machines Essentials, there are just four modelled devices, all based on analogue tape machines. The full version adds more tape machines but also gives us a range of preamp models, to give us 10 flavours in total.

Rather than being named for what they emulate, the virtual devices' names describe their tonal effect and in Essentials, these are Sweeten, Warm, Thicken and Vintagize. An Info button brings up a short description, which focuses on the era of the device, but all you really need to know is that the machines get progressively older, warmer and dirtier as you step through the models from left to right. You then have a single control, Drive, with which to tweak the effect: turn clockwise for more effect. For Essentials, that's pretty much it. But the modelling itself is top-notch, as you'd expect from UA, and between the 'EQ' of the tape machines (complete with head bump, restricted bandwidth and any other quirks in each machine's response) and the saturation (from both the modelled electronics and the virtual tape), there's a surprising range of usable sonic characters here. With so few controls, it's incredibly quick and intuitive to use, and it should appeal to those music-makers who don't want to spend an age tweaking their effects to perfection.



As well as adding more models, the full version allows a hint more control. Again, the models are arranged with the cleanest on the left and the filthiest on the right. The models are Sweeten, Edge, Glow, Warm, Thicken, Vintagize, Distort, Overdrive, Fire and Sputter. Again, the Drive knob determines how much colour they'll add. A second control changes according to the selected model: for tape machines it's a modulation effect called Warble (simulated wow and flutter) and for the preamps it's Tone. Tone gives you scope to shape the overall effect, and can be invaluable for tailoring the sound for a mix (you can do that with EQ too, but it's good having access to this on the GUI when previewing models/presets). Warble can serve up some very interesting results, particularly if using Verve on multiple parts: perfect for wobbly synth pads, texturising layered backing vocal 'oohs' and 'aahs', or generally for building up rich and evolving soundscapes. Finally, you can also set the output level — and I should say that the omission of this welcome but simple touch from Essentials puzzled me.



There are lots of presets to get you started, and a browser that makes it easy to search by the effect you want, rather than the machine used to create it.

The tape machine emulations can certainly sound very cool, but I found myself drawn more to the preamps. Perhaps it's a personal taste thing, but I can see why the preamps are all 'paid extras'! The additional models to those in Essentials begin with Edge, the politest of the preamps on offer here and aptly named: it adds a sort of hard, slightly crunchy edge to the sound that never suffocates the signal unless you really overcook things — reminiscent of a solid-state console preamp. Meanwhile, Glow is a valve preamp, and when you start to turn Drive you're greeted with more clarity and solidity than from any of the tape emulations. It sounds wonderful on rock drums, and can give a nice tight edge to a vocal. The sound dulls a little at more assertive settings, but the Tone control can counter that. Distort is another valve pre but, as the name suggests, it's rather further into effect territory. It offers a different character from Glow, the breakup being both more obvious and softer ('warmer', if you like).

Next comes Overdrive, an old tape machine with Warble control, not much headroom and a very obvious character. It sounds somewhat boxy and claustrophobic, and generally you'll find that it acts as a crude but lovable compressor too, bringing up low-level details and smearing transients. Fire, another tape machine, sounds less vintage, but seems to be pushed harder for much more crunch. Finally, we have Sputter,

a preamp and probably my favourite of the lot, simply because there's so much range in the Drive control that it's easy to dial in an enticing effect. Things seem fairly solid-sounding at the lowest settings, and there's a fairly long and gentle transition as you turn the knob towards, eventually, a very spitty, spluttery breakup.

The minimalist control set is clearly deliberate and generally it's a Good Thing. Still, a wet/dry blend control wouldn't go amiss (Reaper's insert slots have a wet/dry control and I found myself reaching for that often!). The Tone control would have been just as handy for adjusting the tape models as the preamps. And really, I reckon every plug-in should feature both input and output level controls, particularly those costing what these do.

I should also note that there's a little lag on some changes, but this can be helpful depending on what you're trying to do. For example, with a tape machine on an aux send and Warble set about midway, the effect swooshes as the dry and wet signals come in and out of phase. Change that parameter to zero, and the swooshing continues for a few seconds before settling on the new setting. The delay can throw you when auditioning presets, but it's all in sync with the DAW and that means you can use automation or a controller without having to be too precise.

Despite the very gentle caveats above, I have to say that there isn't a bum note in this performance: the models sound every bit as good as we've come to expect from UA, and Verve Analog Machines is so intuitive that you can fire it up and start getting creative straight away. It's not quite like any other plug-in I've used, and well worth checking out. *Matt Houghton*

$ Verve Analog Machines $199. Verve Analog Machines Essentials $99.
W www.uaudio.com

Firesonic FireEQ

Equaliser Plug-in

Available through United Plugins, FireEQ supports macOS and Windows hosts and AAX, VST, VST3 and AU formats, and is authorised with a personal licence key that allows use on multiple computers. FireEQ combines several EQ methodologies with a view to making mixing and mastering flexible but without adding too much complexity. It employs 64-bit internal processing and supports sample rates to 192kHz and beyond.

FireEQ's low- (30-450 Hz) and high-shelf (1.5 to 10 kHz) filters have a ±16dB range and use analogue modelling to emulate vintage hardware, while a single Dark/Bright balance knob can control the overall tonal shading of a mix without resorting to the individual EQ controls. Depth sets the amount of processing and a further knob, Magic, adds complex dynamic processing that evens out the sound while lifting out detail that might otherwise get buried. I suspect there's a lot going on behind the Magic control, as I found it extremely effective.

The low-cut filter is independent of the shelving filters and adjustable up to 500Hz. It includes a 'Sidekill' function (also adjustable up to 500Hz) to make the low bass frequencies mono — this can eliminate the phase cancellation issues that might occur on playback or when cutting to vinyl, and ensures equal distribution of the bass end across both speakers. This filter has a selection of slope options: 6, 12, 24, 36 or 48 dB/octave.

At the centre of the resizable GUI is an eight-band EQ, each frequency (100Hz to 12kHz in octave steps) having its own Side/Mid knob below the level slider. This adjusts the Mid/Sides balance, essentially adjusting the stereo width for each frequency. There's switchable automatic gain compensation that follows the EQ settings to match the before and after loudnesses, so as to avoid the 'louder sounds better' illusion. High-resolution in and out metering, with level adjustment, is available to the right of the GUI.

I found FireEQ to be a very effective 'polishing' tool, and for mastering, the trick of narrowing or mono-ing the low end while adding a hint of extra wideness to the highs can work very well. The Magic knob does an impressive job of adding density while making details more audible, and I like that you can tweak the individual frequency bands and still adjust the overall brightness or darkness with a single control. There's a 14-day free demo version, giving you plenty of time to try before you buy, and once you've bought it updates will always be free. *Paul White*

$ $99 (discounted to $15 when going to press).
W https://unitedplugins.com

Minimal Audio Flex Chorus

Chorus Plug-in

Chorus is one of the oldest BBD (bucket brigade delay) effects, arriving hot on the heels of the first analogue delay pedals. Essentially, chorus is a modulated short delay that adds pitch wavering, and when this signal is added to the dry sound, the result approximates the sound of two instruments or voices playing the same part but with small timing and tuning variations. Alternative designs were created that ran two slightly different modulated delays at the same time, in order to disguise the cyclic nature of the effect, and so the multi-voice chorus was born. With analogue delay chips each new voice adds to the expense, but software emulations can add stereo operation and multiple voices more easily!

Minimal Audio's Flex Chorus, which supports the usual Mac/Windows plug-in formats including AAX, can have »

» up to 24 voices, and offers overall control over the delay time and modulation depth. A single set of modulation controls has adjustable rate and adjustable randomisation, with a sync-to-tempo option as well as a display that shows the left and right delay taps wandering back and forth along a timeline. In the centre are controls for Width, Time and Feedback, with the EQ controls placed to the right and being switchable between Multi-band and Filter types. (Multi-band is designed to preserve the low end and high end of the source sound.) Sliders at the bottom of the window set the input level, wet/dry balance and output level, and a soft-clip limiter can be switched in to catch any accidental overloads. The number of voices can be set to 2, 4, 8, 12, 16 or 24, and while there are no controls for individual voices, it can be seen from the display that the modulation amount, delay times and direction of movement varies between voices for different modulation and delay settings. The modulation is disabled when Depth is set to zero, so that the delay controls can be automated to create custom effects.

Additionally, Minimal Audio have added two operational modes: Smooth and Glass. Smooth is said to work according to human frequency perception, so as to produce a deep, rich sound, whereas Glass mode takes a multi-stage approach for brighter, crystalline effects. There are numerous presets to demonstrate the possibilities, and there's also a random button for help choosing presets when you're in need of inspiration.

Adding more voices uses more CPU, but the CPU load of this plug-in is fairly modest even with all 24 voices switched on. Also, to my ears, using two or four voices gives the best balance between movement and complexity, though using a lot of voices can add density and texture where you want that effect. Having adjustable delay time and feedback allows for effects that veer towards comb-filtering at the shortest delay times, with more noticeable doubling at the other extreme. There's an impressive range of stereo chorus effects available, from all the old guitar standbys and honky-tonk pianos to heavy detuning, faux reverb and shimmering soundscapes that go way beyond standard chorus effects. *Paul White*

$ $49 (discounted to $29 when going to press).
W www.minimal.audio

Warm Audio Warm Bender

Fuzz Pedal

When fuzzboxes first began to appear in the mid-1960s (starting with the Gibson Maestro), UK engineer Gary Hurst came out with his own take on the effect and called it the Tone Bender. His design went through a couple of revisions.

Warm Audio have now built a pedal called the Warm Bender that offers both those versions along with a third 'silicon' transistor setting to give the user three distinctive fuzz voicings.

The first Tone Benders were housed in wedge-shaped enclosures but the Tone Bender was also produced by Sola Sound, who used the pressed metal base of a wah pedal as their enclosure. Housed in a pedal that seems to combine aspects of both the wedge and Sola Sound shapes, the Warm Bender has controls for Attack (drive) and Level but, as with the original, there's no tone control. The input and output jacks are joined by a PSU input (battery power is also supported) and there's a slide switch called Sag that mimics a fading battery by dropping the power to 6V — users of the original often preferred the sound when the battery was starting to die! Another nicety not found on the original is a status LED, as the invention of low-cost LEDs was still a few years in the future at that time.

A rotary switch selects between the three versions. The MkI Tone Bender uses NOS 76 (1x OC76 and 1x SFT337 transistors), and these early Tone Bender sounds clean up well when the guitar volume control is backed off. They can recreate many of the '60s fuzz sounds, though I didn't find a setting quite raspy enough to nail the sax-like 'Satisfaction' sound or Jeff Beck playing 'Heart Full Of Soul' with the Yardbirds, without first adding more treble on the amp. It has lower gain than the MkII but retains the overall tonal balance of the input pretty well and cleans up nicely when the guitar volume is backed off. The next setting has noticeably more gain, more low end and uses NOS 75 (3x OC75 transistors) to recreate the MkII tone. This is a very punchy sound with a lot of warmth. Producing a generally tighter sound, a smooth distortion tone at lower gains and more grit at higher gains, the Silicon setting offers a very useful alternative voice.

Bringing in the Sag control to recreate the dying battery adds a tangible sense of compression in all three modes and when the guitar volume is backed off, it tends to bring in extra harmonics, reminiscent of a subtle octave fuzz. In the case of the Silicon mode, when the guitar volume is backed off and the Sag switch is on, the decay of the notes also tails off in an authentically unnatural way. There's a slight level drop when using Sag too, but the Level control has way more than enough range to make up for that. Overall then, if you're a fan of the Tone Bender era of fuzz sounds, this little pedal covers them all and then some. I sometimes wished for a tone control, but I suppose the aim here is to exactly duplicate those old Tone Bender sounds, and if one was added the pedal would no longer be authentic. *Paul White*

$ $199.
W https://warmaudio.com

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HARDWARE V PLUG-INS

Can plug-ins ever match the sound of outboard and an analogue console? Our unique comparison recreates a 48-channel hardware mix using software alone.

PONTUS 'EVAN' HAGBERG

Hardware versus in-the-box mixing has been a hot topic for almost as long as any of us can remember, but I don't think there has ever been a true A/B comparison. Has anyone completed a full 48-channel mix using hardware, then meticulously replaced it all with plug-in equivalents, matching the levels and carefully replicating EQ curves — not just using equivalent plug-ins, but also using test tones and spectrum analysers in an effort to make them sound, as far as possible, identical? I don't think so. In fact, I don't believe there has been an A/B test of any sort on a full 48-channel mix. Until now: I've done just that, and in this article, I will take you through the process I used and report what at least some of you should find to be pretty interesting findings. But first, it's probably helpful if I explain just how I got to the point where I would be insane enough to attempt such a painstaking comparison...

Hardware Crazy!

I've used hardware by preference for a while. Earlier in my career, though, I mixed exclusively 'in the box', and never quite managed to get my mixes to that point where they sounded 'like a record'. I was happy that I knew what I was doing, and was certainly getting good results. But I still felt that the final product could sound better, so my thoughts turned to hardware — and when I saw a used SSL XLogic SuperAnalogue channel strip in a music store at a bargain price, I bought it. Back in the studio, I used the hardware insert feature in Pro Tools and, lo and behold, my lead vocals now sounded like a record. Whoa!

Of course, that bargain turned out to be my most expensive buy ever! I called the store to ask if they had any other analogue gear in stock and, sure enough, they had a Neve 8801 EQ as well. When I first used that, I almost cried, and my gear obsession then very quickly ran out of control. I wanted more and more channels of analogue processing, and tracked down the last remaining stock of what I consider an extremely underrated product line: SSL's X-Rack, a modular system they created before the 500 series became ubiquitous. The UK prices were fair and the exchange rate with the British pound favourable, so I ended up with five X-Rack chassis, each filled with modules, and during the next 18 months, I tracked down even more units from as far afield as Mexico.

Of course, I needed a lot of A-D/D-A converters to send audio between my DAW and all these real-world channels and, although I'd never really bought into the analogue summing hype (it might change the sound subjectively, but I don't perceive any 'shortcomings' in digital summing), I figured that with all this audio already in the analogue domain, it would make sense to have an analogue summing option available so I could mix in the analogue domain, or pass through fewer stages of conversion. To that end, I ordered a Neve 8424 console (reviewed in *SOS* October 2020: www.soundonsound.com/reviews/neve-8424), a versatile, high-quality mixer with no onboard mic preamps or processing. That was perfect for me, since those bases were already covered by my outboard.

This brings us very close to today, but there was one more step in my productions that compromised two and half albums for me! In recent years, I've been working with a rock project called Symphony Of Sweden, which has had some minor success on Spotify. Even though I was mixing their

tracks in the analogue domain, I felt that my mixes lacked the openness and the airy high end of the best albums. Partway through the third album, I tracked this down to my mastering processor, Softube's Weiss MM-1 Mastering Maximizer plug-in (which uses exactly the same algorithms as the legendary Weiss DS1 Mk3 hardware).

If you're reading this, Softube, don't worry, because I have much better things to say about this plug-in later! But, at this stage, MM-1 was causing real problems for me. Until the point at which I captured the final mixes, everything was sounding great. I used the Neve 8801 EQ and SSL G-Bus compressor on my analogue master bus and then, as the final processing stages, added the Weiss MM-1 and a dither plug-in before bouncing the final 16-bit/44.1kHz master. Part way through the third album, I listened very carefully and could hear that the Weiss plug-in was doing something I didn't like: the energy it added was great, but it also ruined the high end for me. To solve this, I put my trust in analogue gear again: I was one of the first in the world to buy the expensive-but-wonderful LAAL, an analogue look-ahead limiter made by Polish company HUM Audio Devices (reviewed in *SOS* February 2024: www.soundonsound.com/reviews/hum-audio-laal). Half of the songs on our third album *Haunted* have the LAAL in place of Softube's MM-1.

The author's Neve 8424 mixer, with some of his SSL X-Racks above, and (below) just some of the outboard he has acquired over recent years in his hunt for sonic perfection!

Oversampling Awakening

At this point, I was a happy camper: having chosen my analogue path, I could finally make mixes and masters that I was genuinely proud of — that sounded 'like a record' — and I could easily have continued with that approach. But, about a year ago, I was making a sound library for a plug-in/virtual instrument company, and in the process, I discovered something that would make me revisit this analogue/digital debate.

The core idea for this library was to use my analogue synths, but I also wanted to include one of my Firechild trademark sounds from a PPG Wave 2.2. As I didn't have that synth any more, I tried to create it using Waldorf's plug-in version instead. Initially, I thought it seemed nothing like I remembered the hardware sounding, but for reasons I can't really explain — it was just pure luck — I gave it a shot with Pro Tools running at 88.2kHz instead of 44.1kHz. The PPG V3 plug-in came alive at this sample rate. "What the heck?!" I thought. "Is this the solution to *all* my problems?" I tried the same thing with Arturia's range of synths, which I didn't think had sounded great at 44.1kHz, and yes, it worked for these too! Of course, many soft synths already feature internal oversampling and, for example, with Xfer Records' Serum the difference to my ears is huge. But could I apply this 'oversampling' to standard audio plug-ins as well? And might it give me the same improvement?

Back in the day, I had recorded some songs at 88.2kHz, but had found it a hassle. I needed to use double ADAT cables for eight channels, projects took up double the disk space and used double the CPU power, and I was always having to sample-rate-convert third-party audio loops. At that time, the outcome just didn't seem to justify that hassle, and I'd remained on 44.1kHz ever since. But I now had good reason to revisit the issue, and was pleased to find that working at higher sample rates has become much more convenient in the intervening years.

Obviously, lots of plug-ins now feature internal oversampling options, but it turns out that they don't all do a transparent job of it. If you send pink noise through a plug-in and then use a spectrum analyser, you can see in some that the high-cut filter used to eliminate aliasing also brings down the audible high end. Different plug-ins sound better on different oversampling multiples (some offer more multiples than others), but the perfect 'middle way' seems to be 2x. I was a little late to the party, but my experiments led me to DDMF's Metaplugin, which has been a lifesaver. Metaplugin is itself a plug-in, but it can host and chain other plug-ins in a really flexible way. Importantly, it can also »

The sound of many plug-ins benefited from finding the optimum oversampling setting, often with help from DDMF's Metaplugin. Oversampling uses more computing resources, of course, so more use was made of this on critical signals, such as lead vocals and, in particular, the mastering chain.

oversample the whole chain while your DAW still runs at 44.1kHz, which makes oversampling a lot easier to manage.

So, how about that 'evil mastering plug-in' — did oversampling improve the performance of Softube's Weiss MM-1? It sure did. More specifically, when hosting MM-1 in Metaplugin and running at 16x oversampling, it sounded like a dream (I don't know why, but this particular plug-in definitely works best at 16x!). Having spent what I did on the LAAL, I should probably have been crying at this point — but I was definitely smiling! Don't get me wrong, the LAAL is a wonderful analogue processor and it sounds great, but on hearing the MM-1 plug-in do for my mixes what I'd always hoped it could, my world turned upside down in an instant!

The final piece of this revelatory jigsaw fell into place just a couple of weeks before writing this article. I was having dinner at a friend's house, where he has a little music studio in the back yard. I thought of him more as a virtuoso keyboard player (he's the first call for that here in Sweden) than as a producer, but he played some mixes he'd done recently and I fell off my chair. These were 100 percent in-the-box mixes and while he employs some interesting tricks (such as using Studio One's binaural panner to make the mix really wide and spacey, which impressed me), overall his mixes just sounded great.

Though the original mix was done using only outboard, there were several digital effects involved. Among the highlights when it came to the plug-in replacements were a Relab reverb, which did an excellent job of replicating the effect of not only the Lexicon but also the Bricasti hardware, while Wave Alchemy Glow was an admirable stand-in for an AMX gated reverb.

Between that experience, and the results of my own initial tests with oversampling, I had to know if I could mix as well entirely in the box as I could with my collection of analogue gear. So I conceived an insane, hugely time-consuming experiment: I would create an all-digital carbon copy of a complete 48-channel analogue hardware mix, replacing every single bit of hardware with plug-ins, and summing in the box.

The Hardware Mix

The mix in question is a reworking of the Symphony Of Sweden song 'She's In My Head', first released back in 2020. (For this new version we added the word 'Reloaded' to the title.) We used the original vocals but chose to re-record most of the instruments, including the drums, for which, of course, we used high-end hardware, including API 512C mic preamps, and classic 'usual suspect' drum mics such as AKG C414s, SM57s, Neumann KM184s, and so on.

When it came to mixing, I started by performing a full mix with outboard gear only — a 'big console' kind of setup — with the sole aim of trying to create a good, engaging mix. Forty-eight separate channels came out of Pro Tools through my Avid HD I/O converters and they were each routed through an analogue EQ, which, in turn, fed an analogue compressor. Five of these channels had analogue de-essers at the end of the chain (the routing in the studio is hardwired this way) and I used some other gear where the mix required it, including some SPL Transient Designers on the drums. If you'd like to see more detail about precisely what processing was used, check out the chart towards the end of this article. There, I've listed all the hardware I used for the first mix, along with the plug-ins that replaced it in the second.

After the analogue outboard channels, the audio signals were fed to the Neve 8424 console. Though ostensibly

a 24-channel console, this has are multiple inputs, and in this case I routed 24 channels to the console's 24 main input channels, and the rest to the 24 'input C' channels. The former fed the stereo bus directly, whereas the latter were summed by the stereo cue bus, which in turn was routed to the master bus. However, before hitting the C inputs, those 24 signals passed through a pair of Wes Audio ngLevelers (which *SOS* reviewed in March 2023: www.soundonsound.com/reviews/wes-audio-ngleveler). These analogue, VCA-based, automatable level controllers allowed me to perform detailed level automation using my DAW, complete with recall (though for the C inputs only). I completed the mix with this analogue setup as I normally would, though digital processing was involved in the form of my send effects, which included Lexicon, AMS and Bricasti hardware reverbs and two Universal Audio DSP-powered reverbs: a Lexicon 480 and an AMS gated reverb.

The Software Challenge

Once I was happy with my outboard mix, my challenge was to achieve the same result working entirely in the box. I did not want to improve the mix, and neither did I want to prove that analogue was better: all I wanted was to see just how similar I could make the software mix sound to the hardware one. I hope you think I've been meticulous in my approach to this, because it was not easy! It took me a full four days to get this mix right, just like an old-school mad scientist. I stopped only to eat and to sleep (pretty badly!) and at one point I told my wife I was slowly going insane!

When it came to EQ, rather than rely on listening alone, I first ran pink noise through my analogue EQs, and used the Crave DSP EQ2 plug-in, which has a great spectrum analyser, to get a visual 'stamp' of the hardware EQ settings. I set the analyser to its longest averaging time, which made it easy to see just how I should set the controls on the plug-in to match the curve of the hardware EQs. Many channels hadn't required particularly extreme settings, but the drums were a big challenge, in terms of both EQ and dynamics. For the hardware mix's drums I'd mainly used Neve 8803 EQs, but I didn't have an 8803 emulation plug-in, and I chose to use the Brainworx bx_console SSL 9000 J to emulate the Neve settings. This worked very well, although in a few cases (mainly kick, snare and tom close mics) I was surprised by just how extreme the plug-in settings needed to be to match the hardware EQ's curves.

For compression, I found that I could sometimes just dial in identical values on equivalent analogue-modelling plug-ins, and I must say that Universal Audio are really nailing it with many of their plug-ins (the Distressor and API 2500+ in particular really are spot-on emulations of the hardware!). For other compressors, I put the analogue hardware track on the left and an emulation plug-in on a duplicate track on the right, so that I could then see on the meters if I needed to tweak the ratio, attack or release. When I had both level meters on the master dancing identically, I knew I'd matched the compressor settings pretty well (though, obviously, I listened to the effect to confirm that).

The most time-consuming task — but also the most interesting to me, so it's what I chose to work on first — was definitely to pin down the drum sound. Not only did that involve the most radical EQ settings, as I've explained, but I'd also used an SPL Transient Designer on four tracks. There are plug-in versions that do a similar job, but I found that they were just not giving me quite the same 'punch' as the hardware. To compensate for that, I found that I had to be a bit more assertive with those channels' compressor settings. And, of course, as well as the channel dynamics I had to work on the drum bus compression, where I'd used two compressors in series.

The next most time-consuming task was the reverbs. Reverbs are really important for my mixes, so I wanted them to be totally identical, and I happened to use three hardware reverbs on this song. I had to clone two Bricasti M7s and one vintage Lexicon 300 reverb. I also chose to stay away from 'DSP-powered' plug-ins for this test and stick with native processing, so I needed something to fill in for the UA reverbs, too. Incidentally, I'm not a big fan of convolution reverbs for music production, as they don't quite offer the same complexity of texture.

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THE PELUSO P-414 VINTAGE TRIBUTE WITH ITS BRASS CAPSULE IS AN EXTREMELY DURABLE AND VERSATILE MICROPHONE. THE DETAILED MID-RANGE AND SHIMMERING HIGH-END AIR PROVIDE AMAZING RESULTS ON VOCALS, PIANO, GUITARS, AMPS, STRINGS, DRUMS, HORNS...
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» Happily, I found that the Relab LX480 v4 plug-in could match the sound of both the Bricastis and the Lexicon 300. I found that extremely interesting — not so much with the Lexicon 300, since that's 'half a 480', which is what the Relab recreates, but that the Bricastis were *so* close in sound to the 480 was a surprise. (Brian Zolner and Casey Dowdell must have listened to the 480 a lot when designing their M7 algorithms!). As a substitute for the UA AMS gated reverb, I bought Wave Alchemy Glow, which sounded great, and for UA's 480L I just dialled in similar settings in Relab's version.

Mastering Chain

Of course, I still had to replicate the analogue mastering chain, which I usually apply during the final mixdown to avoid unnecessary A-D/D-A conversion. This comprised a Neve 8803 EQ, an SSL X-Rack G-Bus compressor and, finally, the HUM Audio LAAL.

I tried a couple of EQ plug-ins but, again, my focus was not on finding accurate models of the hardware units, but rather on matching the sound. My eventual choice was Plugin Alliance's SPL PQ plug-in, and to match the curve I again ran pink noise through the hardware EQ, first matching extreme settings one band at a time, so as to better understand each band's frequency and Q values, and then backing it off again before replicating the EQ settings I'd used for this track. I'd

Neve 8424 Channel	Instrument	Analogue Outboard	Plug-in Equivalents
1	Guitar high notes	SSL EQ, dynamics (X-Rack: XR425, XR418)	bx_console SSL 9000 J*
2	Not in use	n/a	n/a
3	Intro clean guitar	SSL EQ, dynamics	bx_console SSL 9000 J
4	Intro clean guitar	SSL EQ, dynamics	bx_console SSL 9000 J
5	Verse strings	SSL EQ, dynamics	bx_console SSL 9000 J
6	Verse strings	SSL EQ, dynamics	bx_console SSL 9000 J
7	2600 synth bass	SSL EQ, dynamics	bx_console SSL 9000 J
8	2600 synth bass	SSL EQ, dynamics	bx_console SSL 9000 J
9	Not in use	n/a	n/a
10	Not in use	n/a	n/a
11	Backing vocals	SSL EQ, dynamics	bx_console SSL 9000 J
12	Backing vocals	SSL EQ, dynamics	bx_console SSL 9000 J
13	Evan backing vocals	SSL EQ, dynamics (X-Rack: XR625, XR618)	bx_console SSL 9000 J
14	Evan backing vocals	SSL EQ, dynamics	bx_console SSL 9000 J
15	Hollywood choir	SSL EQ, dynamics	bx_console SSL 9000 J
16	Hollywood choir	SSL EQ, dynamics	bx_console SSL 9000 J
17	Chorus strings	SSL stereo EQ, comp (XR727, XR728)	bx_console SSL 9000 J
18	Chorus strings	SSL stereo EQ, comp (XR727, XR728)	bx_console SSL 9000 J
19	High-pitch BV	SSL stereo EQ, bus comp (XR727, XR626)	bx_console SSL 9000 J
20	High-pitch BV	SSL stereo EQ, bus comp (XR727, XR626)	bx_console SSL 9000 J
21	Lead double 1	XR425, 1176LN, Distressor, SPL de-esser	bx_console SSL 9000 J, UAD 1176LN Rev. E, Distressor, SPL De-esser
22	Lead double 2	SSL XLogic E Signature Channel, Distressor, de-esser	bx_console SSL 9000 J, Distressor, SPL De-esser
23	Chorus lead vocals	SSL Vintage 4000E, Bluey 1176, de-esser	bx_console SSL 9000 J, UAD 1176 Rev. A, SPL De-esser
24	Break lead vocals	SSL Vintage 4000E, Bluey 1176, de-esser	bx_console SSL 9000 J, UAD 1176 Rev. A, SPL De-esser
25+26	Triggered snare	SSL EQ, SSL Bus Compressor (XR727, XR626)	bx_townhouse Buss Compressor
27+28	Overheads	SSL stereo EQ, compressor (XR727, XR728)	bx_console SSL 9000 J
29+30	Not in use	n/a	n/a
31+32	Yamaha CP80	SSL XLogic SuperAnalogue Channel	bx_console SSL 9000 J
33	Kick	Neve 8803, SSL XR618, SPL Transient Designer	bx_console SSL 9000 J, SPL Transient Designer
34	Snare	Neve 8803, SSL XR618, SPL Transient Designer	bx_console SSL 9000 J, SPL Transient Designer
35	Tom 1	Neve 8803, SSL XR618, SPL Transient Designer	bx_console SSL 9000 J, SPL Transient Designer
36	Tom 2	Neve 8803, SSL XR618, SPL Transient Designer	bx_console SSL 9000 J, SPL Transient Designer
37	Tom 3	Neve 8801 channel strip	bx_console SSL 9000 J
38	Solina	Neve 8801 channel strip	Avid EQ1
39	Hi-hat	SSL XR625, SSL XR418	bx_console SSL 9000 J
40	Drum ambience (mono)	SSL XR625, Neve 8424 (talkback compressor)	Waves C1
41	Electric bass	SSL 611EQ, Neve 2264ALB	Sofube British Class A
42	Lower verse lead vocals	Mäag EQ4, 2264-TubeTech CL-1B, dbx 520	Mäag4, Softube British Class A, Tube-Tech CL-1B, SPL DeEsser
43	Rhythm guitar	Elysia XFILTER 500, XR618	bx_console SSL 9000 J
44	Rhythm guitar	Elysia XFILTER 500, XR618	bx_console SSL 9000 J
45	Power chords	API 550A EQ, API 2500,	UAD API 550A EQ, API 2500,
46	Power chords	API 550A EQ, API 2500,	UAD API 550A EQ, API 2500,
47	Power chords verse	RND 551, RND 543	EQ1, Softube Tube-Tech CL-1B
48	Power chords verse	RND 551, RND 543	EQ1, Softube Tube-Tech CL-1B
Mix bus inject	Pro Tools main output	No processing: clean stereo input to Neve mix bus	n/a
Reverb return 2	Triggered drums, drum reverb	Clean stereo input to subgroup 3+4	n/a
Subgroups 1+2	Backing vocals	SSL Bus Compressor, RND 542 (tape emulation)	bx_townhouse bus comp, Softube Saturation Knob
Subgroups 3+4	Drum bus, bass	SSL Bus Compressor XR626, SSL XR728	bx_townhouse bus comp, bx_console SSL 9000 J
Master stereo bus	Mastering chain	Neve 8803, SSL Bus Compressor XR626, LAAL	SPL PQ , UAD SSL G-Bus, Softube Weiss MM-1

*All plug-ins starting with 'bx_' are by Brainworx.

The above table shows the different sources in this song, and the processing used for both the hardware and software versions of the mix.

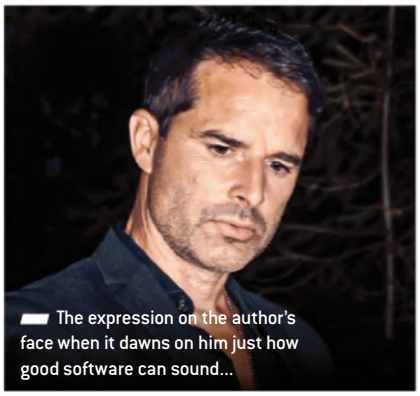
used very small amounts of EQ really, with a 1dB dip at around 280Hz, another 1dB cut around 3.5kHz, and just 0.7dB of high-end boost.

For the G-Bus compressor, I opted for the UAD version and dialled in the same settings as on the hardware: the slowest attack with a 2:1 ratio and auto release (yes, the classic master-bus setting that always works!). Finally, to emulate the LAAL I turned back to the Softube Weiss MM-1 Maximizer. To my ears it sounds a bit like the LAAL, actually, and matching the LUFS level was not difficult at all, but when listening by ear there was a touch of low bass missing, and perhaps some subtle analogue distortion going on in the LAAL too. I looped one bar and used the Crave EQ2 to measure the EQ curve, and found that in order to match them, I had to boost 0.5dB with a 96Hz peak filter. I then dialled in a little saturation courtesy of the SPL PQ's THD feature until I was happy.

To get the right result, I ran the SPL PQ and the UAD G-Bus compressor in Metaplugin set to 2x oversampling, while another Metaplugin instance hosted the Weiss MM-1 at 16x. But while the mastering is kind of close, you can see that as a bonus — my focus for this test was very much on seeing whether I could recreate the mix.

Takeaway Thoughts

To my ears the two mixes sound very similar, and if you'd like to hear them yourself, you can find the files on the *SOS* website at https://sosm.ag/jun24-hardware-v-itb. I was — and still am — surprised, upset, sad and happy at the same time. After all, I invested a lot in hardware after my earlier experiences with plug-ins!



The expression on the author's face when it dawns on him just how good software can sound...

As I've hopefully proven through these tests, when you have a target to aim for you absolutely can reach it when working 100 percent in the box, as long as you take the time to learn your plug-ins and try them with different oversampling settings. Metaplugin really has been a lifesaver for me, particularly for critical instrument and synth parts, but many processor and effects plug-ins also sound better in Metaplugin running at 2x, 4x and sometimes even higher multiples. I also like that when using plug-in chains Metaplugin reduces the number of stages of oversampling (and associated filtering) compared with using plug-ins' built-in oversampling options.

But while I could match the sound to my satisfaction, working in the box was a very different experience from working with hardware. In particular, I found it interesting just how extreme some of the plug-in settings had to be to match what the hardware was doing — this test has given me more confidence to turn those virtual knobs all the way if it genuinely sounds better. (This even extends to panning, by the way: a 'three o'clock' setting on the Neve console's panner sounded almost like maximum right on the Pro Tools pan knob — this probably explains why panning in Pro Tools has previously led me to mix rather narrower than when I mix on the Neve!)

I don't think I would ever have been able to make a mix like this in the box without the prior experience of working with hardware, either. Learning the hardware boxes, and the nature of the physical interaction — being able to dial in the right settings with such ease — has been a great school for me. It encourages you to judge things by ear, without the distraction of what's on the screen. I'd also dare to say that analogue hardware generally has a wider sweet spot, and that means that almost everything you dial in will sound good in a strange sort of way, making it easier to achieve a good mix more quickly.

I imagine that the big question many of you will be asking right now is this: am I now thinking of selling my analogue hardware? No, not just yet. But if my studio were on fire, I would now almost certainly rescue my computer before my X-Racks, and that was definitely not the case six months ago.

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Experimenting with quirky ideas can help to create a fun and creative environment in the studio — and can sometimes provide genuinely inspirational sounds!

Finding Time To Experiment In The Studio

On paid sessions, every engineer has to get the job done. But it's still important to find the time to learn your tools, discover new techniques — and foster creativity.

NEIL ROGERS

We live in an age in which production budgets are tight and every musician who wants it has access to decent recording tools. Consequently, many studio recording sessions are a bit shorter than they once were and they can be a bit more businesslike too: more of a focus on getting work done and less time for experimentation. As an engineer and owner of a commercial studio, I feel I must be reasonably confident that we'll end up with something good before I choose to burn an hour setting up a mic inside a piano that's sent to a Leslie speaker! But it wasn't always this way, and I still feel quite strongly that a recording studio must be about more than good gear, competent engineering and nice acoustics: it should also be an environment that encourages creativity.

In the early days of running my studio, I'd regularly experiment with unusual mic techniques. Things like recording an acoustic guitar in the toilet, or putting mics in a tube or bucket in front of a drum kit. The client would be on board with whatever crazy idea it was, and actively participate in its execution. Of course, the result would often sound awful. But it was nearly always fun and, sometimes, it injected some much-needed energy into the session or provided that spark of inspiration we needed to push the production in an original and more interesting direction. I think it was healthy for clients and artists to see me enjoying and being creative in my work too, and I gained valuable experience of what might or might not work on future sessions. In other words, all this 'play' made me a better engineer!

The Play's The Thing

Fortunately, my reviewing gear for *Sound On Sound* has bought me back some of that time to experiment and play in the studio. When evaluating most equipment, I'll typically start by throwing a device in at the deep end on a real session and then, over the course of several sessions, building up an appreciation of its strengths and weaknesses in various scenarios that are encountered frequently in the studio. But this paints only part of the picture of what a device can do, and it's often only when I start preparing some audio examples (to help our readers get a better feel for what I'm writing about) that more of what a device has to offer is truly revealed.

A big reason for that is that when I'm preparing these examples, it's in a very different setting from a recording session and I'm in a very different head-space. There's no time pressure for a start, and I'm not constrained by the need to serve a particular production or meet a client's expectations. All I'm doing in these moments is experimenting, with no motive other than to figure out what the equipment can do. It's just me, on my own, playing around and observing any meaningful differences made to the audio signals — and often I surprise myself by coming across at least one new way (and sometimes more) of using an EQ or a compressor that I probably wouldn't have discovered if I'd just used it to do a job on a session. For instance, recently, when reviewing a hardware EQ, I found myself boosting and cutting frequencies to what I'd usually consider a stupidly dramatic degree. I was just seeing how the EQ behaved at extreme settings but I really liked the results and, having since bought that EQ, it's a tactic I often now turn to when tracking.

Lots of tools can 'do the job' on a session, but if you take the time to really get to know what your gear is capable of, you'll get more out of it when it matters.

That's just one example of time spent experimenting with gear paying dividends, but whether I'm reviewing a mic, a bit of outboard, a plug-in or whatever, I'm finding that it happens quite a lot. And it's made me much more conscious of the need to make the time to do this sort of thing. In fact, I think that everyone can benefit from carving out specific periods for 'practice' or 'play'. Obviously, this can have a direct benefit in that you get to understand your equipment in more depth, but it also encourages you to develop new techniques that can be quickly pulled out of your hat in recording or mixing sessions. And without it, there's a risk that you get stuck in a rut, growing too comfortable using the same tools and techniques and pushing out similar results time and again — where's the fun in that?!

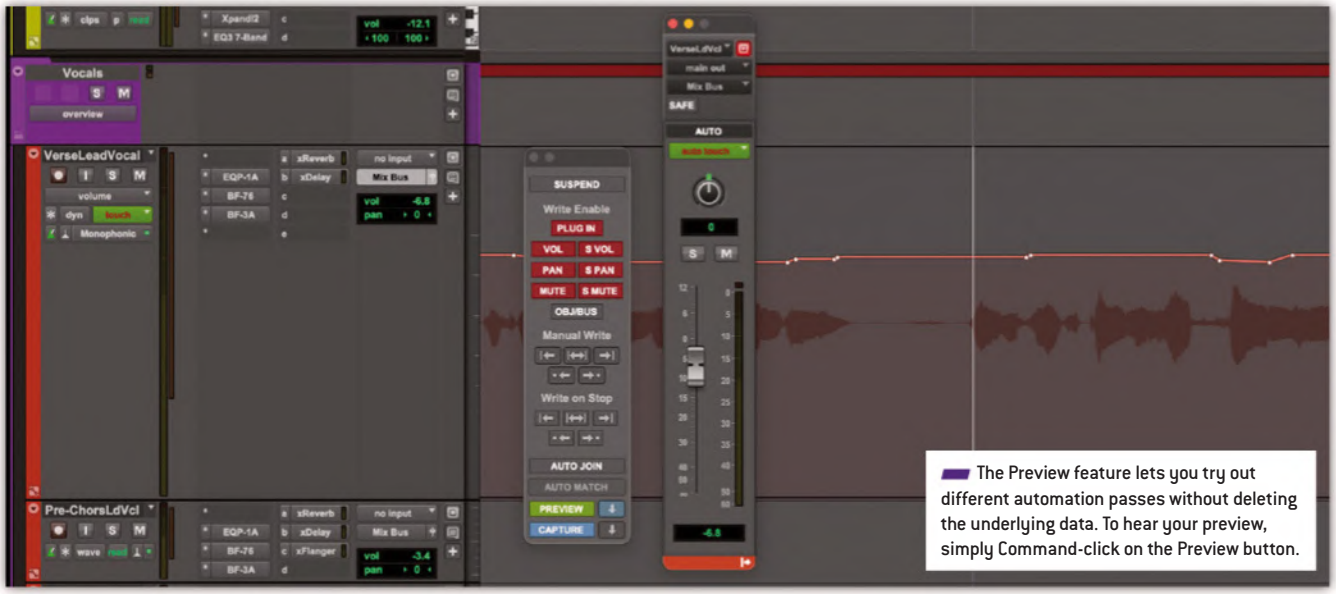
If you're in the early stages of your journey in recording or mixing, then finding the space you need to be able to play and experiment should be pretty high up your list. Yes, you can learn a lot by watching other people do stuff on the Internet, and most of us can play with plug-ins at home too, but finding your own inspiration and having direct hands-on experience is hugely important. This is one of the positives of a well-run audio course: as well as the structured learning environment, the course provider will give you access to facilities in which you can (carefully!) mess around. Alternatively, consider offering to help out with chores at a local studio in exchange for some downtime so you can play on your own time. For more seasoned engineers or those with paying clients, I think it's about respecting the value of the creative aspect of 'play' in studio sessions, and taking the time to experiment for yourself outside the sessions (while still leaning on your tried-and-tested techniques to get results). In an informal way, I've been creating time in my sessions where there's a sense of 'off the clock' fun. Most clients seem to value that and, even with brief periods of play, I get as much out of it as they do.

Even with a great live room to hand, it can help to try different spaces from time to time. (Drummers get to sit on a throne, so why can't guitarists?!)

A Piece Of Work

I suspect, though, that many home producers will actually be faced with the opposite problem: spending too much time playing, and not focusing sharply enough on being productive and finishing projects. Struggling to let go of your creative endeavours as you chase perfection is a well-trodden path to insanity, but with instant recall in our DAW software and the availability of every audio processing tool, avoiding this is arguably even harder today.

While self-discipline isn't the main focus of this article, in many respects it's the flip side of the same coin, and if you start by defining the time in which you *should* be playing then, by implication, you're also setting boundaries that tell you when you need to be more disciplined and your focus should be on being productive. If you find that side of things difficult you're not alone, so I'll leave you with one tip that many people who do this professionally will agree with: it's really important to have deadlines to finish your productions. When your projects have a defined end point, you soon learn to let go and accept, in a positive way, that "this is the best I can do in the time I have." Make a note of any ideas you've had or lessons you've learned along the way so that you can make use of them later. And you know what? It's amazing just how often that project you felt you were being rushed to mix will sound great when you come back to it with fresh ears!



The Preview feature lets you try out different automation passes without deleting the underlying data. To hear your preview, simply Command-click on the Preview button.

We round up some handy shortcuts and time-saving techniques.

JULIAN RODGERS

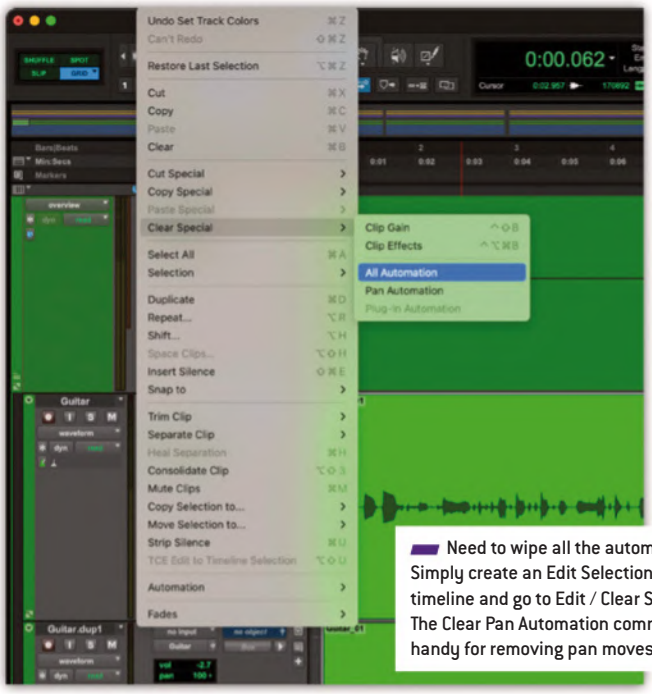
This article was inspired by a recent conversation I had with some fellow Pro Tools users about things that would have saved us a great deal of time if we'd known them sooner. There is no theme to the material presented in this month's article beyond all of these things being very useful, so hopefully there's something new here for everyone.

Certainly in the conversation I was having there were examples where the reaction of seasoned users went everywhere from sagely nodding agreement through to outright astonishment.

I'll start with some useful tips which will benefit you before you even get into your session. These tips use Mac modifiers. PC users should substitute Option with Alt, Command with Control and Control with Start.

Window Pain

We've all had the fun of clicking away notifications and warnings, usually without reading them and often with some irritation. Opening a session in Pro Tools and turning away for a moment while it opens, only to find a warning that something has suspended the process, is an irritation. When you get a whole series of warnings about the same thing, it gets downright annoying. To get rid of the current warning and all subsequent warnings, just Option-click the first one. To get a handle on the circumstances where this applies, just give it a try. You'll soon understand how it works.



Need to wipe all the automation from a session? Simply create an Edit Selection along the entire timeline and go to Edit / Clear Special / All Automation. The Clear Pan Automation command just below is also handy for removing pan moves from Atmos sessions.

I/O Silver Lining

We've all experienced the frustration of opening a session created on a system other than the one we are currently using. If you're presented with silence when you hit Play, there might be a really simple fix, like the output the session is addressing not being available. In that case, a quick trip to the I/O Setup window and selecting an output bus which is available will be all you need to do. However, if things are more complex than that and you just want to get things working, the path of least resistance is just to level the I/O and start again.

The quick way to do this is to open the I/O Setup window, go to the Output tab and hit Default. If you want to nuke the I/O Setup altogether, then just add Option while clicking Default and all of the tabs will be reset to the defaults for the interface you're currently using.

Automation Shortcuts

Automation in Pro Tools is sophisticated and can get complicated. Of the Advanced Automation features of Pro Tools (which used to be restricted to Ultimate but are now available in Pro Tools Studio), one of the most useful is Preview. It's simple enough in theory: it lets you try alternative automation moves in context with underlying automation. It allows you to hear these moves in the proper context, but without going so far as to actually overwrite any automation until you're sure that it's the right choice.

Despite being 20 years old, EQ3 features a very useful Band Solo mode.

My early experiments with Preview were frustrated by the fact that I didn't understand how to toggle between my new automation, which was in the Preview buffer, and the underlying automation. The temptation is to click on the green Preview button to hear your underlying automation, but actually what you need to do is Command-click on that button — this is absolutely crucial to getting to know this very powerful feature.

Another tip, which was completely transformative for me when I learned it, is that rather than switching manually between Waveform view and the different automation playlists, you can go directly to the playlist for any automation-enabled parameter by Command+Control-clicking on the parameter you wish to view. To go back to Waveform view, click the track name with the same modifiers.

Something else I've found myself wanting to do on occasion is level a mix which already contains automation. This can be laborious to do manually, and if it's someone else's mix, it can be difficult to track down all the automation present in a mix in the first place. The quick way to eliminate all the automation on a track, or group of tracks, is to make an Edit Selection along the entire timeline and go to Edit / Clear Special / All Automation. There is a complementary command, Clear Pan Automation, which is particularly useful. When experimenting with Atmos it's not unusual to want to strip out all of the pan automation, which can of course be across several playlists, but to leave everything else in place. This command is what you need at these times.

Final Destination

It is not unusual to want to send audio to more than one place simultaneously. We do this all the time with sends. The point of a send is that it allows you to route audio to an alternative destination at a different level to the main fader. This is very useful when setting up headphone mixes or using reverbs and delays, but there are many occasions when we just want the same level as the main fader, but going to more than one place at a time. Many people use a send for this purpose, and if the send is set to unity gain and post-fade, it fulfils that function. With 10 sends available it's unlikely that you're going to run out, but a neater and simpler solution is to hold Control and route to another output or bus. The output of the channel will go to both destinations simultaneously. This is denoted by a '+' in front of the output label, while ticks in the selector menus reveal the routing. You can route to many output and busses simultaneously if that's what you need.

Plug-in Power

EQ3 is a very old plug-in, but I still favour it over Avid Channel Strip because it offers some features the Channel Strip does not. One feature that they do both share, which is extremely useful, is a band-pass mode. Many people use the 'narrow boost and sweep' method for tracking down specific frequencies for surgical cuts, but a much more efficient alternative is to hold Control+Shift (or Start+Shift on a PC) to momentarily switch to a Band Solo mode, where you can audition that band of the EQ in isolation while manipulating the frequency or Q. These Band Solo features are common in premium EQs nowadays but EQ3 is nearly 20 years old!

If you have a troublesome session which refuses to open, it is very possible that the issue has been caused by a specific plug-in. If you hold Shift while opening a session it will open with all of its plug-ins inactive. This enables you to work through the session figuring out exactly what is wrong. Or, if you just need to get into the session and export some assets without needing the processing, you can just avoid the issue altogether. You can make individual plug-ins active by holding Command+Control and clicking on the plug-in. If you want to speed things up, you can add Option to this combination and activate plug-ins across the whole session row by row.

Combine Pro Tools' Groups feature with its extensive Show/Hide options, and you'll never get lost in a sea of tracks again.

Hide & Seek

There are some powerful features built into Pro Tools for toggling the visibility of tracks, allowing you to control exactly what you're displaying in your session at any time. For example, Memory Locations and Screen Configurations offer powerful and recallable session management features, but sometimes you just need quick on-the-fly toggling of track visibility, and a great way to do this is by toggling by Group.

Mix and Edit Groups in Pro Tools are used to gang together functions across several tracks. The Groups are displayed as a list in the Mix and Edit windows in the bottom left-hand corner of the window. If you Control+Click on a Group in this list you will hide all tracks other than the members of the clicked Group. An All Group is created by default in every session, and Control+Clicking on this All Group can be a useful way to revert to a view of all of your tracks again; however, if your session contains inactive tracks which are hidden, Control+Clicking the All Group will make these visible too, as the All Group contains every track in the session regardless of status. If you like working this way it is a good idea to create an 'All Active' Group to use for this purpose. You can do this very quickly by clicking the disclosure triangle at the top of the Tracks menu and selecting Hide Inactive Tracks, and then creating your All Active Group.

I hope there was something new for you in this list, or a reminder of a forgotten trick. Many other tips and features came up in the same conversation, so expect a follow-up article in the near future!

Custom modulation envelopes can now be constructed more easily in the Sampler Track's Pitch, Filter and Amp panels using the new shape options.

We throw some shapes with the powerful new sound-design options in the Sampler Track.

JOHN WALDEN

The Sampler Track (which is a feature in the Pro, Artist and Elements editions of Cubase) is a simple but undeniably creative tool, so it's great for experimenting with sample-based sound design. It's been with us a while, but in Cubase 13 Steinberg added some new features, including some intriguing spectral-based modes for the sample playback engine. For this workshop, though, I thought I'd explore another of the new options: the shape-based drawing tools now present in the Pitch, Filter and Amp modulation panels. These make it so much easier to design tempo-based patterns, so that risers, pitch drops, patterned filter sweeps and rhythmic volume effects should all feel well and truly in sync with your project.

I want to keep things simple to follow, so with that in mind let's explore some of the possibilities when using just a single sample as a starting point — in this case, I'll be using a sustained single note from a synth pad. There are audio examples on the *SOS* website that you can audition (https://sosm.ag/cubase-0624) and, with the aid of a second sample (a drum loop) and a vocal hook, these include putting these Sampler Track-designed sounds into the context of an initial sketch for a musical project.

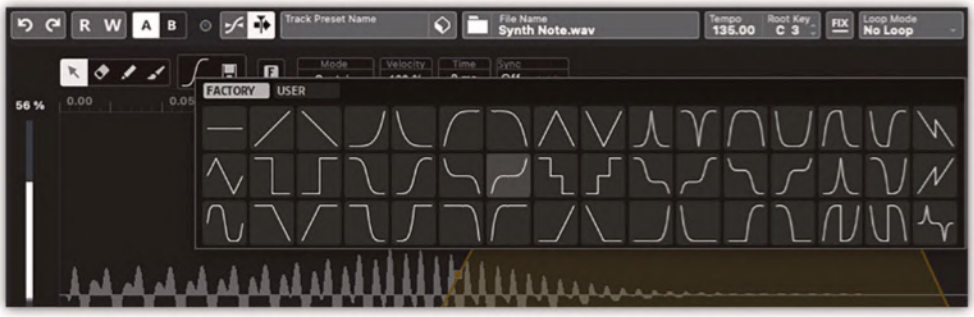
Shaping Shifting

We've explored some of the Sampler Track's established features in previous workshops so I'll not retread that ground other than to remind you that the engine works on a single sample and provides various means of time-stretching (designed for different use cases), loop slicing (for working with drum loops), sample looping (to create sustaining sounds), automatic mapping of the sample across a user-defined key range, and pitch, filter and amplitude toolsets, each with envelope and LFO-based modulation, amongst a bunch of other possibilities.

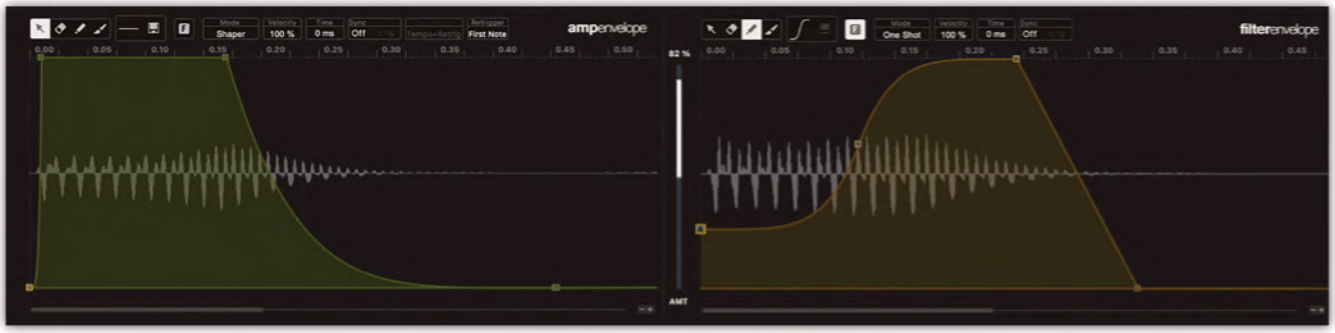
The new 'shaper' tools introduced in Cubase 13 mean more options when creating modulation envelopes. In the Mod panels of the Pitch, Filter and Amp windows, a new toolbar drop-down provides access to a whole range of Factory curves, and you can save your own user curves too. Once a curve shape is selected, you can add it to the modulation envelope using either the Draw (pencil) or Paint (brush) tools. The former lets you draw a single instance of the selected curve, and you can drag as you draw to resize it. The latter lets you add multiple repeats of the selected curve and uses the panel's Sync setting to set the beat division (length) of each repeat (so defining a rhythm for the modulation). By dragging up/down as you use the Paint tool you can control the amplitude of each repeat, and therefore the amount of modulation to be applied. And, of course, you can build custom curves by combining different shapes and/or different beat divisions.

Two other options are worth noting. First, the F button enables/disables Fixed mode. When enabled, adding a new shape (with its curve nodes) replaces any curves/nodes already present in the targeted section of the overall modulation curve. When disabled, existing nodes are automatically shifted to the right along the timeline to make room when new curves/nodes are added. Fixed mode also influences what happens when you edit individual nodes and whether moving one node does (or does not) cause other nodes to the right to also change position.

Second, the Mode types in each Mod panel now include a Shaper



A useful selection of factory curve shape elements are included, and you can also save your own user curve creations.



Even simple curve shapes become easier to generate, as seen here for our simple bass sound with short modulation curves in the Amp (on the left) and Filter (right) panels used to generate a short bass note with a little bit of tonal movement.

option. You don't have to use this mode in order to create modulation curves using the shape presets but it does allow you to create modulation curves that can cycle perfectly, as it ensures the start and end node values are linked.

One Sample, Many Sounds

In creating different sound types from a single sample, we can make use of the Sampler Track's full feature set, but these new shape-based modulation tools can certainly play a useful part. For my (totally arbitrary) mission of generating all my non-drum sounds from a single source sample, there are some obvious contender sounds that I'll need to generate. These include bass, lead, and pad/chord sounds, as well as options such as a risers or bass pitch drops that can be used to add some ear-candy.

Let's start with a simple case: a short (non-sustaining) bass sound. The key settings are shown in the screenshot. In the Playback window, I'd engaged Audiowarp and selected one of the new Spectral modes, then tweaked the Formant and Key F controls to alter the tonality to taste, while in the Filter section I chose a low-pass valve-style filter. However, it's the Amp and Filter modulation that then control the nature of the sound, and each is based on one of the new preset curve shapes. The Amp modulation provides a short, ramped attack, short sustain and then a rapid fade, while the filter uses an S-shaped modulation curve to smoothly but rapidly open the filter as the sound plays to give it a little tonal movement. The Filter panel's AMT control (far left) can be used to adjust the degree of this tonal shift, while the S-curve nodes could be manually adjusted, along with the starting cutoff value, for further control. When played in the lower registers suitable for a bass line, the end result is a short, solid bass tone with a little sonic movement.

A similar filter modulation approach was used to create a lead sound but for this I used a high-pass filter. In this case, the Mono and Legato options were enabled in the Sampler Track's main Toolbar area so that the Pitch panel's Glide control could then be used to provide a pitch slide between overlapping notes.

For this exercise, I created a couple of different pad sounds, both of which involved defining a loop region in the main waveform display (enabling Snap To Zero Crossings and adding a crossfade in/out to the loop region help with creating a smooth looping experience). For the first of these pads, I used a simple curved shape to add a very gentle filter sweep that created a subtle tonal shift as the sound sustained. However, for the second pad, I used the shaper-based tools to create a custom Amp modulation (the main screenshot shows a similar custom modulation curve). When played, this creates a rhythmic effect that's automatically sync'ed to tempo. And, with a further custom modulation curve added to the Filter, I was able to add a tonal element.

The various curve shape options were also useful in creating some ear candy elements. For example, for a bass 'hit and drop', I duplicated my bass Sampler Track and, as well as making tweaks to the filter settings, I added a short pitch modulation curve that smoothly dropped the pitch by an octave. By using the beat divisions in the display, you can adjust the curve shape as you draw it, so its length locks in sync with the project tempo. Similar curves in the Amp and Filter modulation panels can help emphasise the effect created.

I created a number of different risers. Obviously, the initial element required for this is a volume rise that lasts the required time. Again, the beat grid in the modulation panels lets you draw a suitable curve to exactly the length that's required, and I created one-beat, two-bar and four-bar versions using this approach. The rise effect can then be enhanced by using a similar modulation curve for the filter, and I used a band-pass filter this time, to sweep upwards as the riser was playing. In a second version of the four-bar riser, I added a further element by superimposing a small amount of LFO based modulation (based upon a pulse waveform set to 16ths) on top of the shaper-based curve to give the riser an additional rhythmic pulse effect.

Lock Then Load

To keep this workshop easy to follow, I confined my experiments to a single source sample — this is a great way to learn what's possible too! — but in the real world you can use a selection of contrasting source samples to give you a broader range of tonal properties to play with. Also, note that if your modulation curve designs result in something you like, but you'd still like to experiment further, you can try using the same settings with a different underlying sample to produce a variation. To do this, just create a duplicate of your Sampler Track and enable the Lock button (the padlock icon) on the main Sampler Control toolbar strip for the duplicate. You can then load (or drag and drop) an alternative sample into the Sampler Control panel. This replaces the original sample in this instance but retains (locks) all the other settings — including your carefully constructed modulation curves!

As mentioned above, you can hear the results of my shaper-based modulation experiments in the audio examples on the *SOS* website, both in isolation and in a short sketch. It's great that Steinberg are continuing to nudge the Sampler Track feature set forward. No, it's not a fully blown HALion or Kontakt-style sampler, but it's quick and easy to get to grips with, full of creative potential, and well worth digging into if you like to design your own sounds.

Logic's bussing options offer almost unlimited flexibility.

DAVID RICARD

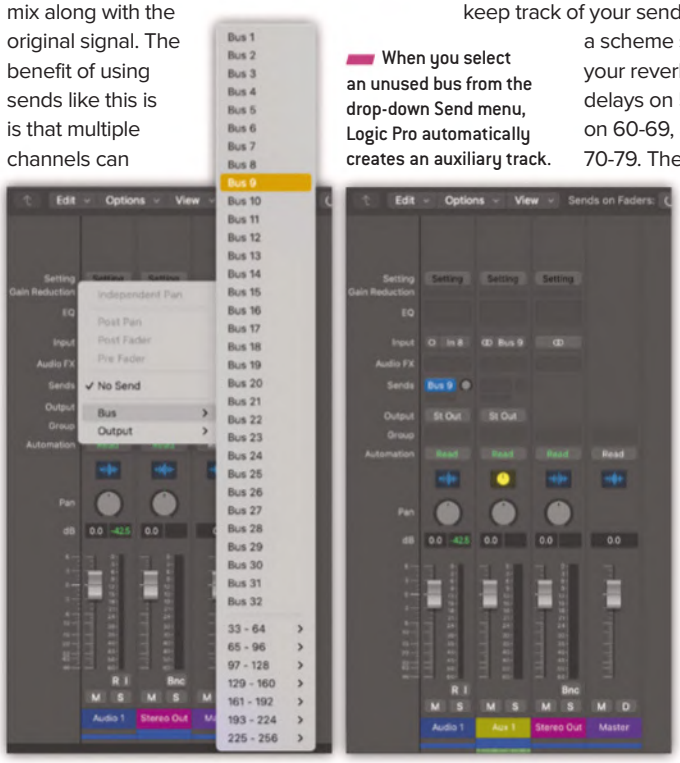
Logic Pro's mixer has implemented busses for pretty much as long as it has been a digital audio workstation (remember, there was a time when Logic was only a MIDI sequencer). With an architecture based on that of analogue mixers, the Logic mixer is intuitive, flexible and robust. But every once in a while I need to remind myself that not everyone has been using Logic Pro for two-plus decades, and that many Logic users aren't dyed-in-the-wool analogue console veterans. In particular, confusion can arise if you aren't fully aware that sends, busses and aux tracks, while related, are three distinct entities in Logic.

Most hardware mixing consoles have several 'aux send' knobs on each channel. Each of these sends distributes signal from that channel to additional outputs on the mixer. During recording, these might feed musicians' headphones. At the mix, they are usually connected to effects units such as reverbs. In this case, the (fully wet) reverb signal comes back into the mixer either through a dedicated effects return or into another channel, and becomes part of the mix along with the original signal. The benefit of using sends like this is is that multiple channels can access one effects unit, but with different amounts of each signal being sent to it. And, since this is how most music was mixed for the last few decades, Logic Pro adopted the same scheme in its own mixer.

Take The Bus

What's left out of this explanation is that the pathway from the send to the reverb is called a 'bus'. In an analogue console, these busses are hardwired to the sends, so when you turn up 'aux 1' you have no choice over which bus you are using. Technically you're 'bussing', but the term is rarely used in this context. In the analogue world, bussing usually refers to sending a number of channels to a group, to be controlled by another 'master' fader.

DAW mixers are infinitely more flexible than their analogue counterparts. So, in Logic, sends are not set in stone as they would be on a traditional mixer: each track can have a different number of sends, and those sends can have different bus assignments on each track. The sends themselves are not numbered *per se*; they take on the numbers/names of the busses they feed. So send 1 on your vocal track might go to a reverb, while send 1 on your guitars could be feeding a delay, and so on. You can name busses, but Logic is unusual in that bus names are global rather than being loaded and saved with each project, so this could potentially cause more headaches than it's worth. If it helps you keep track of your sends, you could stick to a scheme such as having all of your reverbs on busses 40-49, delays on 50-59, modulation on 60-69, and parallel FX on 70-79. Then, by glancing at the bus number, you can easily determine what type of effect you're sending to.



When you select an unused bus from the drop-down Send menu, Logic Pro automatically creates an auxiliary track.

Aux Tracks

Once you've selected a bus assignment for your send, Logic Pro creates an 'auxiliary track'. An aux track is essentially a track with a few minor limitations, the main one being that it is not actually a track. It's a channel in the mixer. In order to have it appear in the Main window, you need to right-click on it and select Create Track (or select it and hit Ctrl+T).

Even if having the aux track in your Main window is not a priority, creating a track for each of your auxes is still worthwhile as it then gives you the ability to move the track anywhere you want, whether you're working in the Main window or in the Mixer. After the aux track is created, you can easily locate it in a busy mixer by holding Shift and clicking on the send slot.

In Practice

Once we have our sends routed to aux tracks, what are our options? Suppose you have a lead vocal track and you want to add some reverb to it. Inserting Space Designer or another reverb plug-in as an insert onto the track would do, but it's actually quite limiting. By sending part of the vocal to an aux track with a reverb inserted, we gain much more flexibility. We can ride the reverb aux track's fader to bring the reverb level up or down as needed, or we can insert a compressor with the dry lead vocal as a side-chain feed, to duck the reverb out of the way on the busy phrases. Additionally, we can add a chorus (or any other effect) to just the reverb.

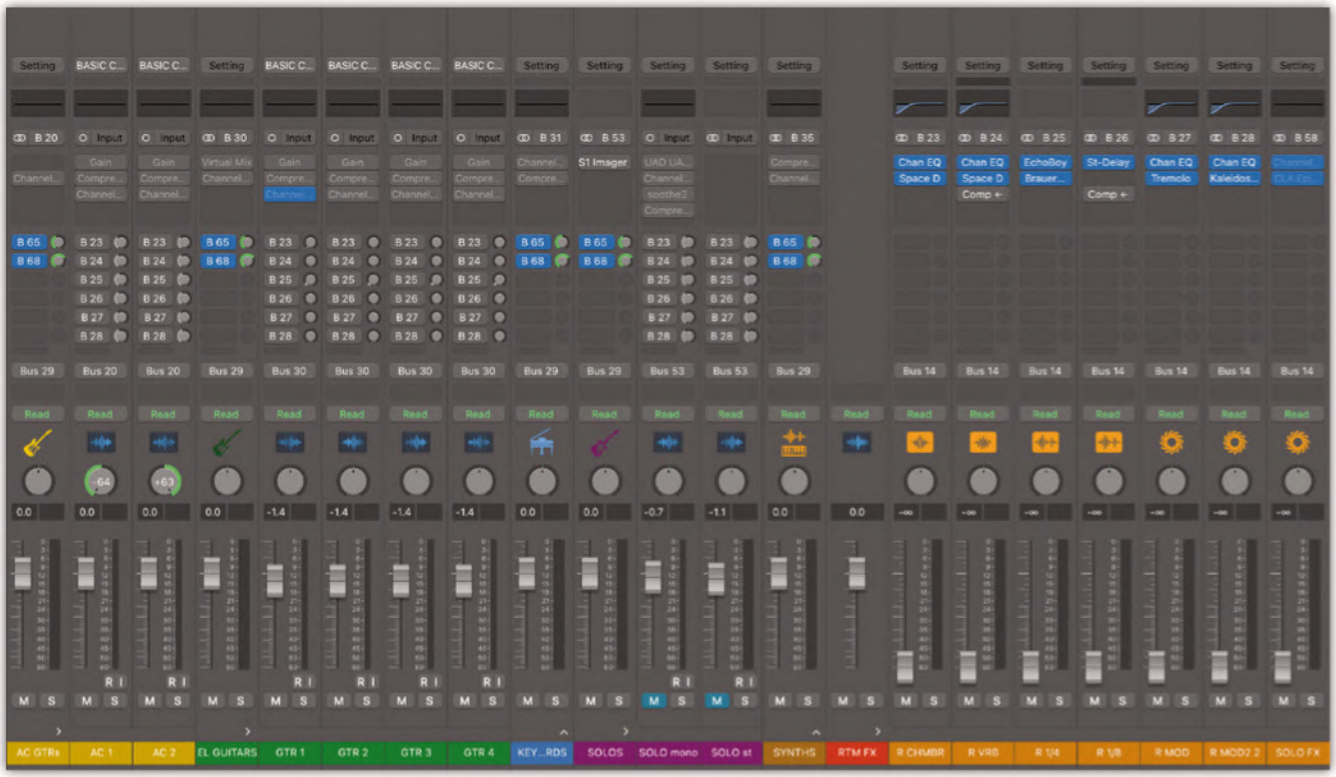
Sends aren't only useful for reverbs and delays. Lead vocals, bass, kick and snare drums can all benefit from parallel processing, and, once again, using sends as opposed to inserting the plug-ins on the original tracks provides more flexibility. For example, I'll often send bass to several aux tracks: one for parallel compression, one for stereo widening, one for distortion/saturation, and one for something out-of-the-box like chorus or delay. Having a dedicated fader for each of these gives me massive amounts of control while remaining incredibly intuitive.

As a word of caution, always check the phase compatibility of parallel-processed tracks with the original dry track!

Beyond The Traditional

So now that we're using Logic Pro's mixer like the analogue offering (albeit with more flexibility), where can we go from here? One break from tradition that I'm seeing — in scoring templates as well as in pop mixing — is a move away from the concept of global effects for a project, and towards creating sets of effects for each instrument group.

In contemporary mixes, it's common for instrument groups like guitars, keyboards,



In my mixing template, I have all of the guitars, keyboards, synths and solos bussed to their own Track Stacks, but have elected to send them all to the same group of effects.

synths, lead vocals, background vocals, bass and so on to all have their own busses. But you can take this further by including all the aux effects for each instrument group in the same bus. The benefit of this setup is that each instrument group is completely discrete, with none of its tracks being sent to effects outside of its group. So, when you turn your drum bus up or down, the drum reverb will follow. This also makes outputting stems much cleaner, and adds an element of control that was previously impossible.

Lately I've been taking the above approach even further. This method works best on an instrument group that typically only has one instrument in it: bass, lead vocal, guitar solo, and so on.

Rather than creating a send and bus for each aux track, I use just one. Start by creating a host of aux tracks (from the mixer menu, choose: Options / Create New Auxiliary Channel Strip, or hit Ctrl+N). Select all of these new aux tracks and set their inputs to a single unused bus. Remember that holding Shift while doing this would assign the inputs consecutively; however, in this case, we want them to all share the same bus assignment.

Now set the bus on the send of the track in question to the same bus and turn the knob to an appropriate level. As you start adding effects to the aux tracks, you can simply use their faders to balance the level as opposed to using the send knob.

Blending Effects

Another potentially powerful application of this concept is to create original effect combinations. Rather than using these aux tracks for different categories of processing (reverb, delay, compression, saturation and so on), why not build a suite of four different reverbs, four different delays, and four different modulation effects? Feel free to insert more processing to each channel and bypass or enable as desired.

Here I have a Lead Vocal track that is housed in its own discrete Track Stack. I have one send feeding a slapback delay, and then bus 49 feeds six channels of reverbs and delays. Each of these tracks has additional processing and optional time-based effects on their own output. The next two sends feed a chorus and a parallel channel. If I decided to add another parallel channel, I would most likely duplicate the channel (provided it was converted to a track first) and change the plug-ins. It's much faster than adding a new send/bus.

Granulator III shows direct recorded audio in red.

We check out Live 12's updated granular synth.

OLI FREKE

Robert Henke, polymathic genius and one of the original designers of Ableton, has contributed a brand-new version of his popular Granulator instrument to the all-singing, all-dancing Live 12. Like the rest of this latest Live, Granulator III sees refinements to workflow and new features aplenty.

For those not familiar, Granulator is a powerful granular synthesizer that processes samples in a very different way to normal samplers, and which often gives rise to completely new and exciting textures. Great for ambient sound designers and anyone who wants to experiment more deeply with sound. It also represents Ableton's ongoing commitment to the Max audio programming language in which Granulator III is written. This new version makes this powerful synthesis method easier than ever to work with.

Those familiar with the prior version — Granulator II (dating back to Live 9) — will notice quite a few differences: gone are the FM, AM and noise sections, the dedicated filter envelope, and the ability to chain two filters together. Not only this, but the remaining parameters are renamed: Pitch Tuning is now called Transpose, but has lost the Fine control that might make it harder to tune patches to a track; FilePos is now named Position; the previously complex Scan section is now just one Scan knob, and Spray is simplified to the less evocative but more descriptive Variation control.

These changes have led to a cleaner, more modern-looking layout, which is in keeping with the Live 12 update in general. There is now a simple row of virtual knobs along the bottom of the instrument, which remain constant, and the top half of the display alternates between the current sample's waveform and the new envelope and filter panels. This makes navigation simple and intuitive.

The real innovation comes with the new modulation options that appear in the narrow row below the waveform display. Clicking a bottom-row knob calls up the modulation options available to it — a different selection appears for each. Alongside standard modulators like LFO, env 2, and velocity, these now include the all-new MPE inputs of Slide and polyphonic Press. I found it easy to check what was modulating what and to quickly adjust settings I had in mind. This simple change has also decluttered the display very effectively.

Vocal Exercise

Sampling your voice is a great starting point to get familiar with granular synthesis: we know what our voice should sound like, so the effect of the individual parameters is immediately apparent.

Use the brand-new direct recording method ('Ext-in' under the 'I/O' button) to record directly into Granulator from a laptop mic. (Other input sources are available, including directly from other Live audio channels — another new feature). Note that Granulator captures the audio retrospectively (which I find a bit disorientating to be honest!) and that you need to choose Channel 1 to record a mono sample if using the laptop mic.

Having recorded your lovely 'ahh', set the Grain size to maximum length to illustrate the differences between the new modes: Classic is the same as Granulator II, and shows two concurrent grains (the cyan vertical lines) moving through the sample. Looping is a new mode, which is more akin to a standard play-through of the sample by one grain. Cloud uses a cluster of grains and you will hear a more chorused version of the sound reflecting this — increasing the Density parameter in Grain Size increases the number of concurrent grains.

Tip: Turn on Hold to avoid the need to repeatedly hit the keyboard to trigger the sound, and tick Loop (in the Scan modulation row) to have the sample continually repeat; left unchecked the scan plays through once and then gets 'stuck' in the grain size window, which may not be what you want.

This is also a useful starting patch to experiment with the newly re-implemented LFO. Click Scan, Grain or Position and control the amount of LFO applied in the new modulation row — and have fun! For super-double fun, alter the Stereo setting in the LFO panel to put the left and right channels out of sync. This can generate anything from subtle to wild stereo spatial effects.

It can be very pleasing to have the LFO applied to the filter alongside one of the other parameters such as Grain. There's just something very nice about that synchronicity! Especially now that there are nine great-sounding filters to choose from, including 'combination' filters to make up for the loss of the two separate filters in Granulator II.

Talking Of Synchronicity...

A significant new feature of Granulator III is the new tempo-sync LFO mode, so I was very interested in exploring the potential for 'granulating in time'! I started with a percussion loop to see if it could plausibly be used for creating variations on a rhythm to bring life and variety to an otherwise static loop. And of course it can! Using the LFO to change parameters on the beat (or other regular timings) leads quickly to very interesting tempo-sync'ed effects that I don't think you could easily create in any other way.

It's a simple case of setting the LFO to the Note setting (as opposed to

the free-running mode) and choosing a number of 16ths under the LFO Beat knob on the bottom row. For example, a setting of 4 will cycle the LFO for the duration of one beat — given four 16ths is equal to a beat.

One slight inconsistency I noticed is that the method of counting LFO time in Granulator III is very different to Ableton's existing LFO Tool. Granulator III multiplies 16ths, whereas LFO Tool uses multiples and divisions of a bar. Not only this but the S&H shape in Granulator's LFO runs four times as fast as the other shapes(!). So a little more consistency in this area wouldn't go amiss.

But once that's understood, loading a percussion loop in Granulator and sending the LFO to any of the likely candidates — Position, Grain, Transpose, Filter — gives instant time-sync'ed results, and as ever with this instrument, a tweak-and-see-what-happens approach is a great experimental strategy. I found I almost always produced pleasingly unexpected and usable results quickly.

Reducing the Shape parameter to below 50 will increase the emphasis on transients for more percussive effects, and running in mono can help increase the clarity. Rectangle and S&H can be used to give very definite changes timed to the beat, and use of Phase determines where the cycle starts in relation to the LFO shape.

Though no-one would claim it replaces a 'proper' beat-slicer it undoubtedly increases the interest and variety of a secondary percussion loop or even a main rhythm. And of course, there's no reason to limit the use of a synchronised LFO to percussion loops — anything timed to the beat is going to be worthy of a try.

Adding Interest To Chord Sequences

Loops can have a tendency to be stuck on one chord, but a neat way of adding granular synth magic to an actual chord sequence (say a fairly static pad sound) is to import a sample of that chord sequence into Granulator III. Then use automation of Position on the arrange page to jump to the correct chord in the sample at the right time in the arrangement. Then experiment with the other settings to impart more texture to the chords.

Summary

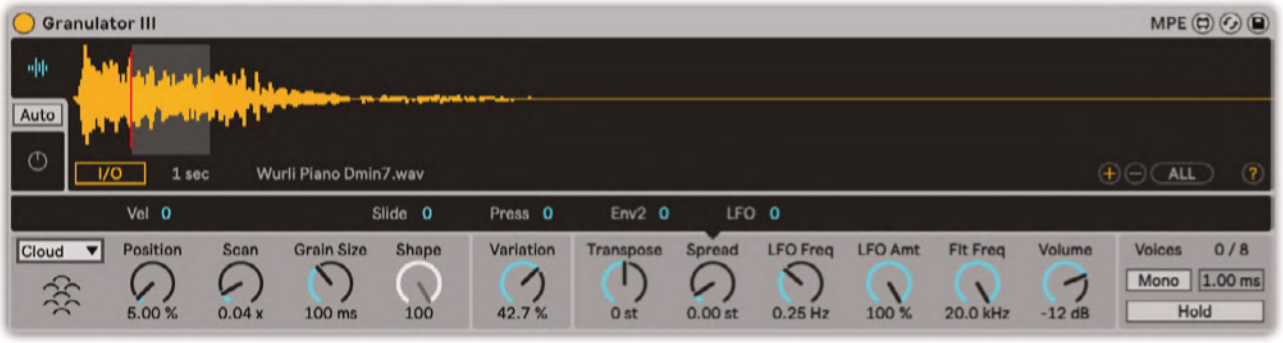
For fans of Granulator II there may be disappointment at the removal of some of the previous features, such as FM, but on balance I think the trade-off is worth it given how much easier this new version is to use — and that the range of sounds possible is still vast.

And given Granulator II is still available (for free), there's nothing actually lost here. But by simplifying and modernising the interface and adding the new MPE integrations, direct sampling and more filter options it is even more inviting and easier to experiment with than ever before. Now there's even more reason to get into the world of granular synthesis and to explore the novel range of sounds that it's capable of!

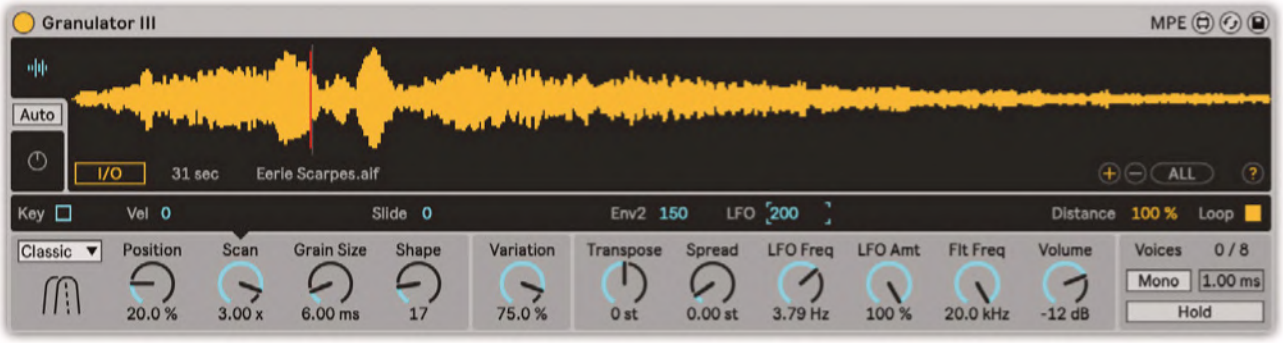
Classic Granular Synth Effects

Granular synthesis reaches the parts of the sonic spectrum other methods can't. Here are a few of the classic use cases that it excels at, with example starting settings:

1. Time-stretched vocals: For the classic 'Fatboy Slim' time-stretch effect on speech vocals set Position to 0 and Scan to 0.1-0.5x.
2. Extend a short one-shot sample into a pad with Scan below 1, Grain Size at 50-100 ms, and use Cloud mode for a richer sound.



3. Long evolving drone textures: the classic granular synth use-case, these rich sounds can be almost be complete pieces of music in and of themselves. Longer Foley or other real-world sounds always work well. Use of Scan and Position with the LFO to jump around the sample is the key here, as is a subsequent rich, wet reverb and delay to create a satisfying drone or wash of sound.



We put the new Lead Architect synth through its paces.

ROBIN VINCENT

I've been reviewing Studio One for *SOS* since version 3, and in every review since, I've moaned about the fact that PreSonus haven't come up with any new instruments since Mai Tai. That was nearly 10 years ago. Well, guess what they snuck into version 6.6? That's right, it's a brand-new and sparkly virtual instrument. Let me introduce you to Lead Architect.

Lead Architect is a sample-based virtual instrument with synth-style filtering, modulation and effects. With a stylish, sci-fi-looking front end and a bit more depth hidden behind the scenes, it has the aroma of one of those Kontakt-based instruments that comes with a large library of sampled sounds and the impression of synthesis. Lead Architect builds its sound through the combination of three parts, each with its own editable synth engine. There are over 140 presets, made from a selection of around 100 individual sound parts. The marketing tells us that the sounds come from sampled vintage synths, field recordings and more.

The main Lead Architect page features at its heart a triangular vector control, with which you can balance the three source sounds.

Presets

The presets themselves are very bold and full of crackling energy. The sounds are high-velocity, dramatic and very, very loud. Designed, perhaps, to knock your socks off with aggressive hits, stabs, strikes and other acts of synth violence, underpinned with dirt and distress. The presets are quite dry and don't have a whole lot of movement going on, but they certainly have an impact and throw you right into the melee if you're coming up with your latest club banger. More liveliness can be found in the Arpeggiator section, and some of those pads are deep and pendulous in all the right ways. I expected a bit more mystery and spacey meanderings, but as a synth for being the architect of leads, it definitely does its job.

The instrument comes with a bunch of 'musicloop' files designed for specific presets. You'll find them under Files / Sound Sets / Lead Architect. If you drag them into your project, Studio One will load the right preset and you have instant lead lines, bass lines and EDM-style loops that you can arrange and edit into your own tracks. They are pretty funky and a great way to get to know Lead Architect.

Front End

On the Global page, most of the action revolves around the translucent triangular panels in the centre of the window. They form a sort of three-sided X/Y pad where you can move a small floating white triangle to create a mix, in a vector-synthesis style, between the three loaded parts. You get one part on each side of the triangle array, and they are colour-coded yellow, red and blue. The triangular sections work really well in guiding you towards interesting ratios between the parts. There are also three small triangles at the points of the inner triangle that essentially solo the part they are pointing at. The triangles are definitely working for me.

One of the part editing pages, showing the synth controls at the top and modulation matrix below.

To the left we have a macro filter control that globally controls the filter cutoff of all three parts. To the right is an envelope that affects the shape of the amplifier envelope for each part, either individually or globally. Both these controls are extensions of the part editors, rather than something applied over the overall sound. They provide some handy quick control without having to page through the parts.

Sample Shift, next to the cutoff, is designed to fool you into thinking you're shifting the pitch of the samples in semitones, except that the pitch stays the same and only the speed changes as if the pitch had been transposed, if you get me. Unfortunately, whenever you click on the up or down arrow it leaps 12 semitones in either direction, and doesn't let you set anything in between. However, if you dig into the automation lanes, you can add an individual Sample Shift lane for each part and change it by semitone there. They do have quite an interesting tonal effect.

Under the main controls is the part selection panel, with all three parts accompanied by the same handful of useful-looking controls. Clicking on the name of the part brings up the part editor page, which we'll see in a moment. The colour coding is really helpful here to keep you aware of which part you are editing. Using the little down arrow you can bring up the part library and select another »

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The Arpeggiator lets you create your own patterns, with editable gate lengths and velocities.

» sound. Next to it is a tiny little padlock that, when locked, only loads the samples for the selected sound, rather than all the part parameters. It's a brilliant way of trying out all the sound design and modulation you've been working on with one sound on a completely different one. Underneath you have a very wide panning slider, and alongside are send amounts for the global reverb and delay. Next to the effects sends is a button that's initially labelled 'FX'. You'd assume this would turn the reverb and delay effects on and off, but no: it actually refers to Note FX, which in this case sends the part to the Arpeggiator or Repeater in the Global FX section.

The Arpeggiator and Repeater functions are really good, and the confusingly named button means that you can turn the Arp/Repeater on or off for each part individually. This can create very dynamic textures, such as bold animated leads over brooding pads. The Note FX are taken directly from Studio One's Note FX and have a comprehensive range of features. The arp can do every direction, with various timings and octave ranges. You can create your own patterns, adjusting the level and gate length per note along with overall velocity and swing. The Repeater not only ratchets notes but can also scale the pitch, scale the gates and be a step sequencer.

The reverb and delay are less impressive. Even with them fully dialled up I couldn't get anything more than around a 60/40 dry/wet mix. It's enough, certainly, in the context of the instrument, but it feels like the sort of synth that could lean heavily into effects. But then Studio One has plenty of other plug-ins you could use for that.

Part Editor

Moving to the Part Editor, we get a modulation section, filter and effects that all seem very familiar. And then you realise it's the synth engine from Presence XT. It's identical except for one tiny difference, which is that the second envelope is called the 'Filter Envelope' in Lead Architect, whereas in Presence it's called 'Env 2'. It's a slightly odd change because the Filter Envelope isn't hardwired to the filter; you have to route it to the filter in the Modulation Matrix, and, oddly enough, in the Matrix drop-down menu, it's called 'Env 2'.

So, we're in familiar territory, which means we already know our way around. We have two LFOs with multiple shapes and a rate that goes from 0.01Hz to 8Hz. The rate can be sync'ed, free or tied to the pitch of the note. The Delay knob is a favourite feature because it can be used to make the LFO come in after a period of time, adding vibrato to the end of a sustained note. The two envelopes have all four ADSR stages, and you can pull out up to 20 seconds' worth of attack, 20 seconds' worth of decay and 30 seconds of release. The 'Filter Envelope' also has a delay feature so you can time it to come in after the note has been struck.

Effects & Modulation

Beneath the editor, we have some effects and the modulation matrix. You can add chorus-type effects, bit-crushing and distortion, along with EQ, panning and a very cool Gater. The modulation matrix has eight slots, in which you can select up to two sources for one destination. So, you could combine an LFO and envelope into the filter cutoff in one slot rather than using up two. You have all the usual physical sources, like mod/pitch wheels, aftertouch, and some MPE inputs from poly pressure and note pitch, along with LFO and envelope outputs. Destinations are a little on the light side. You can modulate every stage of each envelope, but you can't modulate the Drive or Punch of the filter, Sample Shift or any effects parameters.

One thing worth pointing out is that the source material for the parts is pretty great. There's a temptation to be always trying to blend three sounds together, but actually there's a lot of good stuff to be found fiddling with a single sound. Some of the 'Noise Sustaining' recordings are amazing; you could build some brilliant percussive lines with 'Transients', and the individual synth sounds have a lot going for them.

I think the design, intention and general vibe of the front panel is really good. It sets the scene, gives you good information and lets you play with the most important parts of the patch. The triangular mixing system is fun and intuitive, letting you dance around the textures and find the sweet spots within the parts. What's missing is modulation and control. You can't modulate the vector pad. It feels like you should be able to set it spinning or weaving through the space. Maybe it needs the ability to record gestures and loop movements, or perhaps some kind of physics-based system where you can bounce the white triangle around. Grabbing the mouse is the only way to interact with it, which is a bit hampering for a synth. You are able to modulate each part's level in the part editor, but the movement is not reflected on the vector pad. One way around this is to record movements as automation in the instrument's track. Lead Architect will draw three lanes of individual level control over the parts and take those back in on playback, which is reflected in the vector pad. It feels like a missed opportunity.

Conclusion

A lot has happened to software synths and virtual instruments over the last decade, and in terms of modulation and control, Lead Architect perhaps feels a little dated — which is hardly surprising given that it's based on the very mature Presence XT synth engine. At heart, though, it is a thoroughly decent sampled instrument, which has a snazzy front end, some innovative controls, and sounds great.

What Else Is New?

Lead Architect is only available to Studio One+ members, so if you've been wondering why it hasn't appeared in your Studio One update, that might be it. Studio One+ is a subscription service that gives you access to all of the PreSonus software and additional content for a monthly or yearly fee. With version 6.6, PreSonus are introducing Studio One+ Hybrid, which takes over the annual subscription and rewards you with a perpetual Studio One Pro licence at the end of the year. You'll need to continue subscribing to access all the additional content and future full-version updates, but if you don't, you can continue using your current version of Studio One Pro forever.

Also new in 6.6 is the ability to hear how your Dolby Atmos mixes will sound on Apple Music, which takes an idiosyncratic approach to binaural rendering. This includes head-tracking on AirPods, all within Studio One. Sadly, this is only available on Studio One for macOS.

The last new addition is integrated TuneCore support. TuneCore is a distribution platform that can fire your music into streaming platforms such as Apple Music, Spotify, Tidal and so on. You can now launch a TuneCore client directly inside Studio One and upload tracks to your TuneCore account. That will save you a few clicks. If you're a Studio One+ member you can get 50 percent off a TuneCore plan.

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Spotlight Hardware Samplers

The hardware sampler market is healthier and more diverse than ever. What better time to take a quick tour?

Elektron Octatrack MKII

LUKE WOOD

A few years ago, the dedicated hardware samplers that used to populate studios had largely fallen out of favour as their software counterparts gradually became more powerful and less expensive. Now, their popularity seems to be on the increase again — in fact, there are more options on the market today than when we last covered the topic back in 2015. So, this month's *SOS* Spotlight is being placed on a range of hardware samplers, many of which draw their inspiration from the designs of old while harnessing the huge potential offered by modern DSP technology.

1010music Blackbox & Razzmatazz

1010music Blackbox

Despite its compact footprint, 1010music's Blackbox manages to squeeze in a powerful set of sampling and performance features. Recording and editing can be carried out using a touchscreen interface, which doubles up as a set of virtual pads for triggering sounds, while a set of encoders and buttons provide some tactile control over parameters and navigation. Along with level, pitch and loop controls, the Blackbox is equipped with a set of effects, features a granular mode for morphing samples into new sounds, and allows up to 80 samples to be mapped to different key ranges to create custom instruments. As for connectivity, the audio input is joined by three outputs and a headphone output (all stereo), MIDI and clock I/O, and a USB connection that can be used to connect external controllers. If the Blackbox still isn't small enough, the company's Razzmatazz may be worth a look. This tiny instrument combines drum sequencing, FM synthesis and sampling, with each of its eight touchscreen pads able to host two FM oscillators and a WAV file. There's a collection of sounds included, and it's possible to capture samples from the unit's line input before processing them with built-in filters, envelopes and effects.

1010music Razzmatazz

$ Blackbox $649, Razzmatazz $399.
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W 1010music.com/product/nanobox-razzmatazz

Akai MPC

Akai's MPC series has remained immensely popular since its introduction in 1988, and alongside some modern controller variants that require a computer, the range still contains a selection of standalone units. As you might imagine, their onboard sampling and editing capabilities are incredibly comprehensive, but they can also host a wide variety of Akai's effects plug-ins and virtual instruments, turning them into serious all-in-one production tools. All of the current models feature 16 velocity- and pressure-sensitive drum pads and a touchscreen interface, with the main differences lying in the amount of hands-on control, processing power, storage and I/O. The compact MPC One offers four Q-Link knobs for parameter control along with 31 dedicated function buttons, stereo line-level audio and MIDI I/O, and eight gate/CV outputs — there's also the One+, which adds Wi-Fi and Bluetooth, 16GB of internal storage and a stylish red finish. The Live II comes equipped with a similar set of controls but with built-in speakers, three stereo outputs, phono inputs and two lots of MIDI I/O. The flagship MPC X is kitted out with a larger touchscreen, 16 Q-Link knobs, 63 function buttons and seven knobs dedicated to gain, mix and level parameters. The I/O count is increased significantly: four inputs accept mic, line, phono and instrument signals, and there are eight line-level outputs. As for MIDI, there are two inputs and four outputs, and the eight CV/gate outputs are provided on their own TS sockets rather than in pairs

Akai MPC X

via TRS. The MPC X Special Edition takes things even further with increased RAM and storage, and comes bundled with six additional Akai plug-ins.

$ $699 to $2499.
W www.soundonsound.com/reviews/akai-mpc-one
W www.soundonsound.com/reviews/akai-mpc-live-ii
W www.akaipro.com/products/mpc-series.html

Alesis Strike MultiPad

The Strike MultiPad is very much aimed at live performance, sporting nine large, hard-wearing pads designed to be played with sticks. A staggering 8000 sounds are included, and there's 32GB of onboard storage that can be used for user samples, which can either be imported as WAV files or recorded directly from mic and line-level sources as well as via USB. Sounds can be edited after being imported or captured, with the all-important start- and end-point options joined by parameters such as pitch, normalise, reverse and more. There are two stereo outputs with configurable routing — useful for distributing a metronome separately from the drum sounds for live performances, for example — which are joined by MIDI in and out/thru connections, and the unit can be expanded with up to three external drum or cymbal triggers, a hi-hat pedal and a pair of dual foot controls.

$ $599.
W www.soundonsound.com/reviews/alesis-strike-multipad
W www.strikemultipad.com

Elektron Digitakt

Elektron's Digitakt is a combined drum machine and sampler that pairs eight audio tracks with eight MIDI tracks that can be used to sequence external gear. It comes loaded with over 400 factory samples spanning electronic and acoustic drums as well as synth sounds, and sports a stereo input that can be used to capture mono samples. Each audio track benefits from two filters and two LFOs, as well as an amp envelope, overdrive effect and independent delay and reverb sends — there's also a stereo compressor that can be used across the master output. Along with its 64-step sequencer, the Digitakt features a Song Mode, which makes it possible to create, edit and play compositions that utilise dozens of patterns, offering a fast and intuitive way to add some variation to live performances. The day before our press deadline Elektron announced Digitakt II, which gives you all of the above but with more memory, more tracks, longer patterns, stereo sample support and all sorts of other extras and improvements.

$ $849.
W www.soundonsound.com/reviews/elektron-digitakt
W www.elektron.se/en/digitakt-explorer

Elektron Octatrack MkII

Elektron's flagship sampler offers eight audio tracks that boast individual lengths and time signatures, and can host samples captured from the device's four audio inputs. There's also plenty of modulation on offer thanks to three LFOs per track, as well as a huge collection of effects including filters, chorus, phaser, flanger, delay, reverbs and more, any two of which can be applied to each audio track simultaneously. There's no shortage of hands-on control and a wealth of performance features are present, including a powerful crossfader function that can be used to adjust multiple parameters at once or morph seamlessly between different sets of sounds. The Octatrack also comes equipped with comprehensive looping, sequencing and mixing facilities, and can easily be used as part of a larger setup thanks to MIDI in, out and thru connections.

$ $1599.
W www.elektron.se/gb/octratrack-mkii-explorer

Isla Instruments S2400

Inspired by E-mu's iconic SP-1200, Isla Instruments' S2400 aims to combine the sought-after sound of vintage samplers and drum machines with the flexibility of their modern counterparts. A 16-bit/48kHz Hi-Fi sampling engine is joined by a 12-bit/26kHz Classic option that employs hardware anti-aliasing filters — something that formed a key part of the SP-1200's distinctive sound. The device's eight channels each feature a velocity-sensitive pad, a 60mm fader and a rotary pot, along with mute and solo buttons, and can be loaded with any sample from the device's 32 slots. The S2400's sequencer offers a range of different programming modes and includes per-step configuration of all parameters (level, pitch, envelope, filters) as well as per-track swing and quantise options, and also features 32 MIDI tracks that can be used to trigger external instruments. There are four line-level and two stereo phono inputs, along with a stereo mix output and eight independent channel outputs, MIDI in, out and thru connections, and a USB host port that will accept external MIDI controllers. There's also a built-in audio/MIDI interface that makes all of the unit's outputs and a stereo input available to a computer.

$ $1799
W www.soundonsound.com/reviews/isla-instruments-s2400
W www.islainstruments.com/product/s2400

Polyend Tracker+

Just as we were going to press we got wind of the Polyend Tracker+, an updated version of their popular sequencer and sampler that should be available by the time you read this. Tracker+ builds on the concept introduced with their original Tracker — which drew inspiration from early sequencing software — adding more processing power and memory, stereo sample playback and recording, 14 tracks of audio over USB and five synth engines. Now with a track count of 16 — eight sample, MIDI or synth tracks plus eight synth or MIDI tracks — the onboard sequencer supports patterns between one and 128 steps, which can each contain a note, an instrument number, and two effects (a wide variety of choices ranging from volume automation to LFO modulation are available). Not to be left out, the Tracker Mini will also receive a 2.0 firmware update to include the new synth engines and the extra MIDI and synth sequencer tracks, making it completely compatible with the Tracker+.

$ $799.
W www.soundonsound.com/reviews/polyend-tracker
W www.soundonsound.com/reviews/polyend-tracker-mini
W www.polyend.com

Roland SP-404 MkII

The latest iteration of Roland's SP-404 sampler has been treated to some significant upgrades since its release, with the most recent firmware adding the Loop Capture mode from the discontinued SP-555, multitrack audio exports, waveforms for generating melodic content, and more. Despite being a fairly compact device,

» there's plenty of hands-on control: you get 17 pads and a healthy selection of function buttons, as well as four knobs dedicated to volume and key parameter controls, with an OLED screen providing some helpful visual feedback. As well as being able to record samples directly to the pads, the unit comes loaded with a collection of factory sounds, and also boasts a range of built-in effects, some of which can be applied to input sources as well as loaded samples. As for connectivity, the SP-404 features stereo line-level inputs and outputs along with an instrument/mic input and MIDI I/O, and can also capture or stream audio via USB.

Roland SP-404 MkII

$ $499.99.
W www.soundonsound.com/reviews/roland-sp-404-mkii
W www.roland.com/global/products/sp-404mk2

Roland SPD-SX Pro

Roland's flagship percussion pad is aimed squarely at live performance, and comes loaded with 1550 samples that range from acoustic and electronic drums hits to synth sounds, loops and effects. Users are free to capture their own samples via line-level inputs or a built-in USB interface (which also offers MIDI and the unit's audio outputs) and assign them to the pads, where they can be edited in detail using the onboard display; parameters such as pitch and volume can be edited in real time from dedicated controls, with more in-depth settings available via a menu system. It's possible to add to the unit's nine pads thanks to a range of external pedal and controller inputs, with a fully kitted-out system offering a total of 19 triggers. A main stereo out is joined by four assignable mono outputs, and MIDI in and out/thru connections are provided for integrating other gear.

$ $1199.99.
W www.soundonsound.com/reviews/roland-spd-sx-pro
W www.roland.com/global/products/spd-sx_pro

Sonicware Liven LoFi 12 / LoFi 12XT

As its name suggests, Sonicware's Liven LoFi 12 aims to deliver a retro sound that matches its appearance. It is built around a 16-bit, 12/24 kHz engine that features a 12-bit sampling mode designed to deliver the gritty, lo-fi sound associated with early hardware samplers. The rest of its features are decidedly more modern, though: each of its four sequencer tracks benefit from independent effects (of which there are 12 types) and Parameter/Sound Lock functions that can record control movements and sound changes into a pattern either live, or via step-by-step programming. Sonicware have also recently announced the LoFi 12XT, a more powerful version of the device that utilises the technology developed for their SmplTrek unit. It is available to pre-order at the time of writing, and should be shipping by the time you read this!

$ Liven LoFi 12 $239, LoFi 12XT $399.
W sonicware.jp/pages/liven-lofi12

Sonicware SmplTrek

Sonicware Smpltrek

Designed to equip users with the tools to create music anywhere, Sonicware's SmplTrek is packed into a compact, portable enclosure, will happily run on batteries, and even features a built-in mic and speaker. It can still form part of a larger setup, though, with full-size MIDI I/O joined by sync I/O, four quarter-inch TRS sockets that provide a stereo line-level output, and two inputs that accept line, instrument and dynamic mic signals. The unit will also function as a USB audio/MIDI interface. There are 10 sequencer tracks that can be used in five different modes (Loop, Shots, Drum, Instrument and MIDI), along with three Global tracks that host longer recordings such as vocals or instrument parts. In addition to its sampling facilities, SmplTrek comes with a library of over 500 pre-made sounds, and plenty of processing is on hand thanks to 36 insert, send and master effects. Despite its compact footprint, there are 15 velocity-sensitive pads that can function as drum pads or a keyboard, as well as a collection of function buttons, parameter control knobs and an OLED display.

$ $429
W www.soundonsound.com/reviews/sonicware-smpltrek
W https://sonicware.jp/pages/smpltrek

Teenage Engineering EP-133 KO II

Teenage Engineering EP-133 KO II

Teenage Engineering's take on the retro sampling experience comes in the form of the EP-133 KO II, which builds on the capabilities of their hugely popular PO-33 KO, adding more power and extended sampling capabilities along with a new sequencer and effects system. A collection of samples and loops come pre-loaded, and the unit is capable of capturing audio from either a line input or built-in mic, which can be chopped into shape manually or automatically and assigned to one of the 12 pressure- and velocity-sensitive pads. The device has been designed with ease of operation in mind, and the sequencer promises to provide users with a fast and intuitive way to try out new ideas, with a Commit button allowing successful ones to be captured instantly. Among its collection of effects are 12 'punch-in' processors, which can be assigned to buttons and respond to pressure, making quick work of creating filter sweeps, stuttering effects, pitch-bends, and so on.

$ $299.
W www.soundonsound.com/reviews/teenage-engineering-ep-133-ko-ii
W teenage.engineering/products/ep-133

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https://sosm.ag/aro-part-2

Dead Slow

MSM: Producing Casisdead

Casisdead's *Famous Last Words* may be one of the best British rap albums ever made — but actually making it was an epic challenge!

SAM INGLIS

"When it first came out, I was surprised that I could listen to it," says MSM. "I thought my PTSD would have been stronger!"

The 'it' in question is Casisdead's debut album *Famous Last Words*, which MSM was initially hired to mix. His role gradually expanded until the project took over his life completely — for eight years. "It was the most intense project I've ever been on in my life. I don't think there'll ever be an album that could ever top it for me. I was recording, mixing, editing, changing sounds, calling people in to do things. There was always a layer of: 'Can it be better?' in the back of our minds.

"I reckon I tried to quit the album at least five times. I remember being like, 'I literally cannot do this any more.' On one of the songs, I did 500-plus mixes. Not different mixes as in sonics, but comps, vocal edits, changes with breakdowns. I mixed the whole thing, but that happened as we were making it. I'd mix a song, think it's all done and good and gravy, then he's gone to re-record the vocals. And I'm trying to explain to him the amount of automation work I've done, and how it doesn't just work like chopping it in and replacing it. I have to redo all that work. Those are the parts that sent me crazy."

Casisdead's perfectionist art is as much visual as it is musical.

Behind The Mask

MSM (Michalis Michael) would eventually be credited as Executive Producer on the album, more than a decade after he first became a confidant of the secretive rapper. "I knew of Cas from when we were younger, he was just a local MC. Castro Saint was his name at the time. I was a fan, but didn't meet him. Then he just kind of vanished from making music.

"In 2012 I got a phone call from a friend of mine, and he said 'Have you seen this?' The video was what got me first. I was like: whoever's done this is cinematic. Someone's thinking. I recognised the voice, but the name was Casisdead. I've rung around a few

Michalis 'MSM' Michael in his North London studio.

people that I knew and I was like, 'Is this who I think it is?' And they were like, yeah. And then somebody said, 'Oh, he's a good friend of mine. I can I can put him in touch with you if you want.'

"I didn't know he was this MI5 secret squirrel. But he jumped on a three-way call with us and you could tell he was hesitant. He was just like, 'Who are you? What do you want?' And I was like, 'I just want to help. I just want to be involved.' I was working out of Britannia Row at the time and we just met up and played music to each other. From there I set up a studio in my house, and he'd started to transition from the place that he was working to mine. And I was basically picking up the pieces of what he'd been working on, which was a mixtape, *The Number 23*.

"So he was like, 'Look, I'm gonna get you all these files, do what you can with them' — and they were a mess. It was like MP3 instrumentals, vocals recorded on different days with different mics. It was a salvage job. But that's also what I was used to for like nine or 10 years before that. It's not what I wanted it to sound like, because now I've developed, but I know I can do it. And I was just interested in it. I wasn't gonna be a snob and be like, 'Unless you get me the parts...'

"We carried on working, and then around 2015 we did a song called 'Pat Earrings'. When they brought me the demo of it I knew something was special. I heard it and I was like 'Well, this is good. I want to work on this but now I've got the access to parts. We're gonna change this for the better.' And I think by now he'd learned to trust me. So he was like, 'OK, let's go for it.'

"The before and after in that song is one of my proudest things because it became the start of the 'exec production' thing with him. From what it started as compared to what was finished was night and day. But the idea and song was sick and was always there. It just needed to be pushed in the direction Cas wanted. Once it was released it just started going everywhere. It was getting sync'ed in places, there was a lot of labels talking, and then XL [*Recordings*] picked him up. And then from 2016 to end of 2023 we were working on what became the album..."

MSM: "Mike Skinner gave me his Anamod tape simulator. I said 'What!? Why are you giving me this?' And he's like 'I'm not using it. If I ever call you from jail, just answer the phone.'"

Coming Together

"I didn't even know it was going to become an album until a few years in. We would just go to the studio and make songs. We'd put together a setup for him too, because I realised that he's so particular and pedantic about how his performance is. He's a true perfectionist. I couldn't physically carry on mixing for everybody and maintain the recording relationships, so he got a vocal chain consisting of a Neve 1073, 1176 and two microphones of different flavours along with a set of [*Neumann*] KH310s for monitoring. I said to him 'Look, you should be able to record these vocals at home.' He would send me ideas and there was not a day that I didn't speak to him. It was a constant thing. 'Here's something new, what d'you reckon?'

"Since 2016, this is what I've known in the background. I think the reason I stuck it out the way I did was a testament to myself, to say 'I don't give up. I'm not going to quit anything.' But also I respected his work ethic. He seemed to enjoy the art of perfecting things. Some of it was neurosis. Some of it was him being mental, in my opinion. Like I thought he was overthinking some things and I would have to be like, 'Bro, relax. No-one can hear the difference between these two words. This is a decision that you need to make on your own and commit to it.' He would commit to it — and then a week later change his mind again because he could get it better. And did.

"It got to a point where I was in Cyprus on holiday and I was basically having a panic moment, because he wanted something doing and I didn't have my rig. I rang a friend of mine, Felix Joseph, to get involved and he ended up producing a few of the songs on the album. I was like, 'Bro, I really want you to help me on this album.'

"Initially, he was gonna do exec production on it because he's more of the writer and producer than I am. They made some of the best songs, in my opinion. But Felix's work is very different to mine. He takes on sessions that maybe would last a week, two weeks, whereas »

MSM accumulated a lot of high-end analogue gear during the making of *Famous Last Words*. He is also, by his own admission, obsessive about headphones!

» I'm doing a mix a day, two mixes a day. There are gaps in my projects quite often, especially when it's in a single-based culture. We all kind of fought quite a bit about who's doing what, who's picking up the pieces, and at a certain point Felix was done writing and producing and sent me all the files... So I stepped in and was like 'Right, I'm gonna pick up where I left off.'"

Grand Plans

"When I started the album, I was giving it all the big talk and saying 'We're going to use all analogue. We're going to be summing everything.' Mike Skinner [*The Streets*] gave me his Anamod tape simulator. I said 'What!? Why are you giving me this?' And he's like 'I'm not using it. If I ever call you from jail, just answer the phone.' And I was like fair play, that's a good deal!'

"I didn't do the analogue side of life just for my sonics. It was more to encourage him. It was to give him a feeling of like, 'We're using all the proper stuff.' It was an exec producer trick that kind of backfired on me, basically, trying to lift his morale while mine was getting robbed. I rang Cas. I was like, 'Bro, I've got this tape simulator, we're gonna use that on all these songs!'

"I don't know how I feel about it at the moment, but I learned a lot about what I love. I looked at these boxes as things that would get me way closer to 'Let me get the mix out the bloody way, because I know we're about to spend two months comping, changing, editing.'

"The album started with an analogue master chain. All the lead vocals were getting printed through things. I ended up getting a few [*Chandler*] Curve Benders at one point, because one was just glued on a certain couple of settings and I needed another one to experiment with. When I say it to myself now, that's crazy. It has stepped controls, so you can just put it back where it was. I just couldn't be bothered, because I was losing my mind.

"The Avalon 747 hasn't moved since 2016. It's just set. It was used on the mix bus on quite a few songs. And on stems. I just liked this one setting one day and I was like, well, because of the way the sliders are, the risk factor in moving them is ridiculous. I did send sweeps through in Room EQ Wizard so that if someone does move it, I can match it.

"A lot of his album had similar boxes used. There was a lot of 747, there was a lot of Curve Bender, SSL Fusion, Lavry Gold [*MkIII*] and a lot of Anamod. The SPL Iron didn't end up being a mix bus processor, it ended up being a vocal processor along with my Phoenix. They're my two favourite vocal compressors as of late. Very expensive for vocals, but... It just does something. It's like an energiser. It gave life to things."

Mix & Match

Central to MSM's approach to using analogue gear was his SPL Hermes mastering router, which allows different chains of stereo processors to be stored, recalled, and used in parallel configurations. "I'm sending out, on an insert in Pro Tools, into this. Then you choose the box that you want it to go on. Then you have a wet/dry, which is analogue, and then it prints back a stereo stem. So, a lot of the time I'll do drastic

The SPL Hermes mastering router is key to MSM's distinctive approach to working with analogue gear.

EQs on things like a Curve Bender. Even my NTI here, you could use a super drastic setting, almost distorting slightly, but then you just blend five percent of it in with the original signal.

"Pultecs were on kick drums most of the time, but printed. I used to run them through this Hermes and wet/dry it, just to bring in some real bottom end on a kick that he had. I'd print it back in and he always used to say to me 'How did you get so much life out of these poxy drums I've sent you?' I know that works. The only thing I need to worry about is how much of it I add in to the dry signal, print it and move on.

"There was a period on the album where I had two master busses, and all my auxes that would normally be going to my summing mixer, I would have them going to two outputs. One would be going out analogue, one would be going to an in-the-box mix bus. And so I would then flick between the two mix busses comparing the two, like: is analogue summing doing anything for me? That was fascinating. I also did that with different A-D converters, changing the input as I mixed to see which I vibed with.

"The process is where the difference happens in summing. I think of it like this. You can go to a restaurant that Gordon Ramsay runs, and you've given him shit knives versus good knives... both meals in the end are going to be great, but ask him how he felt when he was making the meal. If I'm gonna spend eight years on an album, let me enjoy it, please. Let me have some fun!"

MSM kept records of the complex analogue processing chains for each track in Session Recall.

The success of *Famous Last Words* was reflected at this year's Brit Awards, when Casisdead was voted Best Hip-hop/Grime/Rap Act.

Back In The Box

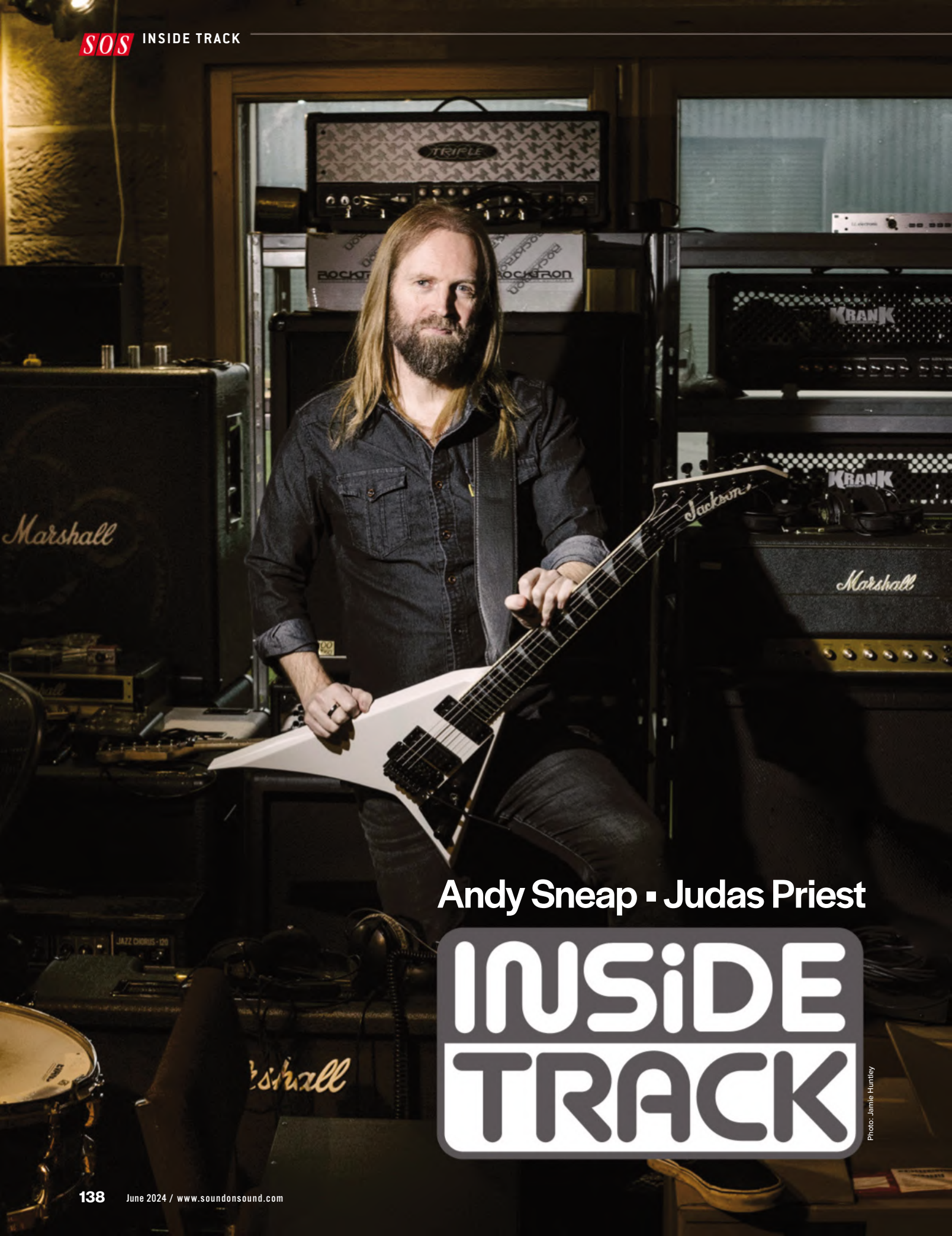
As the revisions mounted up, though, the analogue approach eventually became a millstone. "That kind of died out, I'd say, three-quarters through the album. I was like, 'No. The last couple of songs we're gonna do in the box, and we're gonna use this as an experiment to see if you can match them to the other ones and have it cohesive.' I still took his vocal and would print it through things, but it wasn't all being summed and it wasn't all going through an analogue chain. And that's literally because, if we're changing the vocal comp after the mix so many times, it just was too much for me, even in the amount of time it takes to live bounce five versions of it before you go home.

"I could have just programmed Bounce Butler to do all of this and not have to worry. But I like to see that everything is exactly where it is, because he's so particular. He could hear the difference in mixes. He pulled me up quite a few times, like 'Version 1 and version 3, something's different in the kick.' And I would come in and I'm like, 'Yeah, he's right.' You can't bullshit him, so at a certain point I was like, 'Let me just go in the box and then there's no room for this error whatsoever.'

"Because he's hearing in minutiae. The only other person I know that can hear like that is Dappy from N-Dubz, but his thing is more timing of vocals. Cas can hear flies fucking. He hears the two mixes and he's like 'Version 3 does not sound like version 1. What have you done?' And I've got to come in here and be like 'Why the fuck did I do analogue? What am I doing?' His ears are annoying!

"There was one day when I begged my manager, give him the money back for what we've started! We argued so much. Like there was points where we were not talking. No-one knew who's responsible for what. At one point Felix handed the project back to me, let's say, and my approach was to just match Cas' work rate. So what I would do was like, if he had a change, I would do the change instantly. I wouldn't question it. I would send him the two versions. I'd be like, 'Right, you've got to let me know by this time.' I'd put the pressure on him, so it was like passing this hot potato constantly between each other. And then, towards the end of it, we realised that we're allies in this. And it kind of got a bit more like, 'Do you know what? We've got to finish this. Otherwise we've just wasted years. Let's, you know, not kill each other.'

"I didn't know I was exec producing when I started. I fell into that by the fact I just kept finishing things and doing things. If you'd have asked me at the end of last year, I'd have said never again. But these are the projects you're proud of the most. The reward for me is stuff like seeing him win a Brit Award after all this."



Andy Sneap ▪ Judas Priest

INSIDE TRACK

Photo: Jamie Huntley

When the opportunity came to work with his childhood heroes Judas Priest, Andy Sneap didn't just engineer and produce — he also mixed, mastered, and became the band's touring guitarist.

PAUL TINGEN

Andy Sneap is one of the leading producers and musicians in the metal genre. In a career spanning more than 35 years he has been involved in the making of over 100 albums, by bands like Megadeth, Dream Theater, Hell, Kreator, Opeth, Killswitch Engage, Saxon, Testament, Exodus, Overkill, and countless others. Even so, when he was asked to work with Judas Priest, it was a childhood dream come true. "I got into metal when I was 11-12 years old. I had Judas Priest posters on my bedroom wall! They were one of the big metal bands in the world and I loved them!"

Sneap engineered, mixed, mastered and produced Judas Priest's *Firepower* (2018) and this year's *Invincible Shield*. "In 2016, Mille [*Miland Petrozza*] of the band Kreator suggested to [*guitarist*] Glenn Tipton that I produce their next album, and after some discussions, the band asked me to produce it with Tom Allom, who had produced almost all the band's classic '80s albums. We had a really good time in the studio making *Firepower*. Then, just three weeks before the US tour started, in March 2018, they asked me if I was willing to step in for Glenn. I wasn't going to say no, and here I am, six years later, still doing that!"

Bringing It All Together

Firepower was released on March 9th, 2018, and was hailed as one of Judas Priest's strongest efforts since the '80s. Just before its release, Tipton revealed that he had Parkinson's disease and would no longer tour, hence the last-minute request to Sneap to step in. The *Firepower* world tour lasted until the Summer of 2019. Work started on *Invincible Shield* later the same year, but its release was delayed by more than four years due to the pandemic and the band's 50 Heavy Metal Years anniversary tour, in 2021-22.

"We started *Invincible Shield* with writing sessions at Glenn's place with Rob [*Halford, singer*] and Richie [*Faulkner, guitarist*]," Sneap recalls. "So Richie, Glenn, Rob and I were putting the song ideas together. My role was similar as with *Firepower*. They bounce their ideas off me, and I put the demos together in Pro Tools and make them sound as much as possible like finished songs. I was programming drums using Toontrack's EZdrummer and Superior Drummer. We ended up with about 12 ideas that we were confident about.

"I remember us in February 2020 talking about Covid. We thought that it would all blow over in a month, and that we'd soon be on the planned tour with Ozzy, and cracking on with recording the album. Of course, everything ground to a halt for a long time. As soon as we could, we got back into a studio for the album recordings. But because of pandemic restrictions, the sessions took place all over the place.

"I went to Nashville, where Richie and Scott [*Travis, drummer*] live, and we had 10 days to record his drums in a small studio in the area, called The Southern Oracle. After that, Richie laid down his rhythm and lead guitar parts at his home studio, The Falcon's Nest. He sent me the DI recordings, which I re-amped at my Backstage Recording studios in Derbyshire.

"I recorded Ian [*Hill*]'s bass parts in hotels during the 50 Heavy Metal Years tour, and did two vocal sessions with Rob in early 2023 at Premier Studios, Phoenix, Arizona. Glenn's parts were recorded at his place. I mixed and mastered the album at my studio from August until October 2023. I was doing Accept and Saxon albums at the same time, going between these projects all the time."

Drum Tracking

"The live room at The Southern Oracle in Nashville has high ceilings and a concrete floor, so it is a lively space. The studio has an SSL XL-Desk, with 16 500-series units. With regards to mics, we had a bunch of stuff to try. sE Electronics sent me a couple »

Metal pioneers Judas Priest have been active for more than 50 years.

Scott Travis lays down a drum take at The Southern Oracle.

Photo: Jeremiah Scott

» of drum packs, and I had mics from those on the toms and kick. I had a couple of Crimson Audio-modded Shure SM57s on the snare, two Shure KSM32s for overheads, and some of the sE mics as spot mics. The room mics were Lauten, who had sent me some of their bigger condensers to try. I had a couple of United 47s in front of the kicks. They're pretty good 47 [*FET*] clones.

"Richie and I were there to track Scott's drums, because many of the arrangement ideas were Richie's. As I mentioned, before Covid we had the demos pretty much locked in, with programmed drums. We wanted Scott to put his stamp on it, so it was a case of sitting down with him and working over ideas, especially that Richie had. Scott would spend a day going over a couple of the songs, do his homework, and then we'd come in the next day and he'd bash out the final ideas, usually within three takes.

"Dealing with these recordings during the mixes was fairly straightforward. I added some samples, and then there's a lot of what I call 'donkey work' of cleaning a kit up, particularly the tom mics, so we don't have spill all over the place. I'm trying to get things to sound tight sonically. With this sort of music, you want to have the clarity in the kit and especially make sure the phase in the overheads is correct. That drives me nuts. As soon as you start filtering overheads, your whole phase can change. Especially if you've got a lot of spot mics, like I did, because of the amount of cymbals Scott is using."

The Down Low

The reason for recording Ian Hill's bass parts on the road was, says Sneap, "because we've got so much downtime when touring. So Richie and I sat down with Ian every afternoon in my hotel room and recorded the bass, using my Pro Tools rig with the UAD Twin Apollo, going straight in. When recording we obviously have to use something for playback that's not too heavy on the CPU, so we don't have too much latency, especially when using a native rig. I'll use something like the SansAmp plug-in. Later on, back in my studio, I'll really dial in the tone with the UAD Ampeg B-15N Bass Amplifier plug-in, which I love. I've used that over and over again on many records, and it always works well with Ian.

"Sometimes I re-amp the bass, but I didn't bother on this record. I tried it on *Firepower*, because I've got a decent bass rig back at the studio. But I wasn't getting anything that I felt gave me a real advantage over the plug-in. Later on, during the mix I roughened up a few bits with the SansAmp, and compressed the low end with the FabFilter Pro-MB or Waves C4. I'll tend to use the C4 before any bass amp sim, so it's clamping down before it hits the amp, and I then compress it again. I also often use the UAD SSL EQ, boosting around 900Hz-1kHz with a sharp Q, to get the bass to stick its nose out a little bit. You can trick the ear into thinking that the bass is louder in the mix that way. You get the aggression from the pick attack, but it does not get in the way of the guitar riffs."

Keeping It Real

Richie Faulkner's guitar parts were laid down by the lead guitarist himself at his Falcon's Nest studio. "I think Richie was running a Marshall head into one of those Mesa Boogie Cab Clones, and he was recording DI at the same time. He then sent me the DI, which I re-amped at my place, because he doesn't have the same facilities to track guitars.

"I sometimes use amp sim plug-ins, because some of them sound really good. I'm not a purist about this. I've been using Kempers for years, from even before they released them. We used a prototype on Accept's album *Stalingrad* [*2012*] and I was blown away by it. Things like the Kemper,

the Quad Cortex, the Neural and STL Tones plug-ins sound great. I'm not opposed to them at all. All these people who say they can tell the difference, no they can't. It's really close these days."

For *Invincible Shield*, Sneap preferred to, as he explains, "use real amps going into real speakers, recorded with real microphones, to give us the tones we wanted on the record. I've got the original birch-ply Marshalls from the '80s loaded with Celestion Vintage 30 speakers. I'm always trying to find good vintage Vintage 30s, because the year they were made makes a difference. I've got one or two cabs that I know sound really good. One thing that a lot of people don't realise is that changing your cab makes easily as big a difference as changing the amp.

"I try and have two very similar setups now for left and right, when I'm doing two guitar tracks, but not exactly the same, because that makes the sound a bit smaller if you're not careful. If there are any faults within the tone, it just exaggerates it. For the rest, I keep it simple. I used to be a one-mic guy who finds the sweet spot and places one SM57 there. On the Priest record I had a combination of a 57, an Austrian Audio OC818 and an sE VR2 ribbon on each cab. I found that the mix of those three worked really well. It got rid of the fizz of the 57 and added some extra body and resulted in a better balance.

"It's still a case of dialling in the sound you want on the amps, and then moving mics by less than an inch trying to get the phase to a sweet spot where the scratch goes out of the speaker — going off centre a little bit with the 57, maybe bang on centre with the 818, and finding a good place phase-wise for the ribbon mic. I don't do too much EQ on the way in. Again, I keep it simple. It really is about balancing the mics to a point where they give me what I want in the mids and in terms of being aggressive. I also add a little bit of a room mic in there. Fifteen, 20 years ago, I wasn't really bothering with room mics on guitars, but I've come back to adding a bit of space and air. You can't really hear it, but you feel it.

"Richie and I talked all the time while I was re-amping his parts. We live stream to each other when I'm doing that. I know roughly what he wants, and he trusts me. With *Firepower* we worked together at Glenn's, but because of the pandemic and the 50th anniversary tour, we couldn't do that this time. We constantly went backwards and forwards on this stuff. If he hadn't been happy with it, we would have found another solution. And at the end we had a couple of days together at my place when we patched bits up we weren't 100 percent happy with."

When it came to Glenn Tipton's contributions, says Sneap, "We did all recordings at his place. We'd work through parts when he's feeling good, and we then later pieced bits together. He's got a lot of ideas, so he'll often work with Richie and explain to him what he's hearing, and Richie will put these ideas down. There was a lot of this when we were doing demos, and we used bits from these demos for tracks like 'Vicious Circle'. We also used bits of his rhythm playing on the two songs that came from the *Firepower* sessions, 'Sons Of Thunder' and 'Giants In The Sky', that were co-produced with Tom Allom. I re-amped these parts to get the right tones. I didn't play any guitar on the album, so Rich is doing the bulk of the playing. Glenn was more of an overseer and gave us his opinions and his ideas on arrangements and harmonies. He was very present in the creative process."

Trust Your Ears

In the beginning of 2023, Sneap travelled to Phoenix for 10 days of vocal sessions with Rob Halford. "During these first series of sessions with Rob, we got most of the main vocals down. We were doing a song a day, after which I'd go back to the hotel room to comp what we had. When we »

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» came back the next day, we'd work on the song a bit more, and we'd start on the next song. I was careful not to overwork Rob, because obviously, as soon as the top end starts going out of someone's voice, you're wasting your time.

"Two months later I went over again, and we did all the backing vocals, and worked on the mid sections to the songs and tried to make them work with the solos that Richie put down. It was quite a long process, but we got there. Rob had harmony ideas, and I'd compile those, and we double-tracked vocals to make the choruses bigger. I learned from Tom Allom that the Priest vocal trick is to get the minor third harmony, sung an octave below. It gives it the *British Steel* sound. The second sessions were a lot of putting the fairy dust on. Rob did seven, eight, maybe 10 takes, and then I'd comp them, and I'd try a few different approaches. I'll always double the choruses so we can spread the vocals to make them sound bigger. I double-track the harmonies as well, so I can spread them out of the way of the main vocals, making sure they're still the focal point.

"For microphones on Rob, I used the Austrian Audio 818 for this album. On *Firepower* I had an Advanced Audio C12, but in this case I travelled with the Austrian Audio because it's not too big, and it ended up beating the mics that we had in the studio. I don't have to worry about tuning much, Rob is really on the money with that. Also, many plug-ins, even Melodyne, won't pitch aggressive, edgy vocals. So you really have to trust your ears. Many singers, when they do the high stuff, have quite a wide vibrato, perhaps covering a whole note. It's a matter of taste of where you think the centre note is falling.

"When tuning by ear, I'll turn the volume right down, because if something's out of pitch a little bit, it'll be jumping out of the mix. If it's not jumping out the mix and not catching my ear, it's fine with me. I also think you don't have to be pitch perfect on everything. It just has to feel right. I don't tend to pitch high vocals in any case. They are more about performance and getting it to sit right. Other than that, I may use SynchroArts VocAlign for a big choir or something, to nudge things timing-wise. Especially with the t's and the s's it can get messy."

Hybrid Mixing

When Sneap started mixing *Invincible Shield* at his Backstage Studios in August 2023, he still had quite a lot of work to do.

Priest & Pro Tools

Andy Sneap's involvement with Judas Priest is the culmination of a long journey, which started with posters of the band on his walls. He left school at the age of 15, and his first band, Sabbat, released three albums on Noise Records, 1988-91, with Sneap writing most of the music. However, the guitarist discovered very early on that he felt most at home in the recording studio. "I found recording fascinating. I had a small Tascam Porta One with Sabbat and was always putting demos together. I loved doing gigs, but it was the songwriting process and putting recordings together that really fascinated me. I found the creative process in the studio really inspiring. And recording Sabbat's first album in a proper studio in Germany, live off the floor, was mind-blowing.

"By the time Sabbat split up in the early '90s, I had an eight-track reel-to-reel in the rehearsal room, and I was demoing my own stuff and other bands. I really enjoyed that, and so fell into the engineering and production roles. I never had any training. I made mistakes and learned from them. You're turning knobs thinking about what you want to hear, and then you learn how to actually do it. It was a very natural route to me."

Sneap was an early adopter of digital recording. "I started working with Pro Tools in 1998. I bought a blue and white Power Mac G3 and a Digidesign ADAT Bridge. I had a Mackie desk and five ADATs, because two of them would always be in repair. I was using Pro Tools purely as an editing tool back then. I was recording to two-inch tape, and sync'ing that with Pro Tools, flying things in, editing, and then flying them back. I think it was still Pro Tools 4.3.1. There was no grid, no MIDI. It was very basic. We weren't really thinking of it as a multitrack. It wasn't until 2001 that I started doing full albums in Pro Tools.

"But already with my ADAT setup, I was quite happy with the sounds I was getting. I always thought that digital got a bad rap because it meant musicians could afford to record themselves, and so everyone would blame the equipment rather than themselves. If they couldn't get a good tone, it was always, 'It's digital!' I never found that. I got some great guitar tones on ADATs. True, those early Digidesign 882 and 888 interfaces didn't sound great. But some tape machines didn't sound too good either. I remember spending an hour every morning lining the tape machine up and cleaning the heads. I don't miss that in the slightest. Most of these people that say 'tape sounds better' probably never worked in a studio with tape."

What Andy Sneap does miss about the days of tape is the approach to recording that it necessitated. "There definitely was something really nice about working with tape, because the band had to develop the song more and rehearse more before coming in, and often played the songs live in the studio. Whereas today you're working off a template a lot of the time. Half a year ago I did an album with Myke Gray, who used to play in Skin, and he wanted to record 12 songs in two days, like the old BBC sessions, with three takes for each song. The vibe was really good, because it was actually a band playing. And we left small mistakes in.

"I think we've all fallen prey to looking at the grid and correcting too much, whereas it's the mistakes that add to the feel of a recording. I certainly was guilty of this 15 years ago when Pro Tools was taking off. I don't over edit things now. I don't cut and paste rhythm guitars anymore. I make the musician play through. If it's cut and pasted, your ear recognises it and it gets boring. I try and make the musicians work a little bit more in the studio now. I try to get a full performance, rather than something that's compiled."

As well as occupying the producer's chair, Andy Sneap has stepped into Glenn Tipton's shoes as live guitarist with Judas Priest.

Andy Sneap's Backstage Studios is home to an impressive array of amps and cabs. These were used to re-amp the guitar parts on *Invincible Shield*, with a combination of sE, Austrian Audio and Shure mics.

"I'll go through everything in the sessions. I spend a lot of time cleaning the tracks up, and getting rid of anything that I don't want. I organise the sessions so they're always the same, starting with the kick at the top. I clear out any tracks that aren't used. I'll then have stems that I route through my SSL: kicks, snare, stereo toms, overheads with hats and ride in, and drum reverbs coming up on 'miscellaneous'. Then I have bass, rhythm guitars, overdubs, lead guitars, vocals, backing vocals, keys and miscellaneous.

"I have an SSL AWS desk with 24 stereo channels, which gives me easily enough for what I want to do. I've got probably 16 to 18 stereo stems coming back up the SSL, which gives me a little bit of analogue control. I tend to mix within the box and then bring those stems up, and I can then reach for the SSL desk EQ. Fine-tuning in analogue adds another five percent. I wouldn't be upset if I didn't mix through the SSL, but it's a better workflow for me. It might be because I came up working on SSLs. Going back through the desk gives me a bit of that old-school feel of hands-on approach. I like to see everything in front of me. If I want to push up a guitar at the last moment, I can just grab the fader, and it's there, rather than scrolling through screens. I also use the SSL automation after Pro Tools.

"My other hardware includes Focusrite interfaces. I have a couple of the old 192 Digidesign interfaces as well that I use as insert points. I'm still running Pro Tools as an HDX rig with the cards. It's not native, but it works. I'm still on Genelec 1031s and Yamaha NS10s with a sub. Together with the SSL, it's stuff that I grew up using. I was always on an E Series SSL with NS10s and a sub. I also have some outboard that I tend to use on the way in, and very occasionally in the mix. I've got some Neve mic pres, and some old dbx compressors, and some of the AudioScape stuff that I really like, plus old LA-2As and 1176s, from different makes. They sound great.

"With this album, I tried some of the outboard in the mix, but I wasn't hearing any advantage, so I ended up using a lot more UAD stuff. The simplicity of recalling in the box also is a consideration these days. I'll use quite a few plug-ins in the box. There'll be lot of the Metric Halo Channel Strip on my drums, which I have been doing for 20-odd years. I think that's the best-sounding non-SSL SSL plug-in you can get. The compression is great. The Boz Digital Big Beautiful Door plug-in is great for overheads. I use that to get the snare out of the overheads a little bit. The UAD stuff is fantastic, as are the FabFilter plug-ins, and the Korneff stuff.

"For mastering and compiling I actually use PreSonus Studio One. I love it. I used to use WaveBurner, which came with Logic, but that got incredibly buggy once Apple took over. Then I started using Studio One because I know the guys at PreSonus. They've been great because I've suggested stuff to them, little shortcuts and things, and they've added them in. So I print a stereo mix in Pro Tools that I then export to the Project part of Studio One. You can basically compile the DDP files and burn your stereo WAVs for vinyl and do all the separate digital tracks as well. It's well thought-out.

"I always use the FabFilter Pro-L 2 limiter for mastering. It's great. I've got a setting in that that I always use that is really transparent. I tend to master quieter now than in the past. During the loudness wars, 10-15 years ago, we were all pushing things to the limit, but now I think if you're at about -8 LUFS with rock music, you're in a safe space, where you have enough energy, but you're not crushing things, you're not losing the dynamics. When we were mastering to tape back in the day, it gave us an extra push on the final mix and a bit of glue. If you're careful with your limiting today, you can do the same thing. You want the snare to sit back that fraction and make sure it's feeling like everything's sitting together."

TalkBack

Eduardo De La Paz

WILLIAM STOKES

Growing up in the western Mexican city of Guadalajara, Eduardo De La Paz didn't have many opportunities to learn the craft of recording and production. Playing in bands growing up, he says, "We would scrape our pocket money together for one day in a studio in Guadalajara. I'd be the one pestering the engineer, asking, 'How do you do that?' Mexico has moved on a lot, but at the time there were like, two studios in my hometown. In Mexico City there were way more. So my options were either to go to Mexico City and knock on doors, or go somewhere else."

That choice eventually led De La Paz to London, working under vaunted mixing engineer Craig Silvey on records by the National, Arcade Fire, New Order and more at West London's Toast Studios. He has since made his own name as a formidable mixer, in the process founding offshoot studio Tostada.

At the moment I can't stop listening to

I mean, the honest answer is probably kids' music, because I have a child now! But for pleasure, it's Nadine Shah. Her new record, *Filthy Underneath*, I think is amazing. I was a big fan of her anyways, but this new record is just a step up. Ben Hillier in production. He's amazing. I'm a superfan. And the songwriting, with all she's been going through... It's super raw. Super honest. Especially this last month, I've just had it on repeat. So yeah, when I'm not listening to 'Baby Shark', that's my answer.

I'm currently working with an artist called Attawalpa, who is doing a TV show with his wife, And one of the scenes is just someone playing at a bar. They asked Nadine Shah if she would do it, thinking, "She's just gonna say no." And she was like, "Yeah, I'll do it!" She seems very up for just helping musicians, anyone who needs help.

The project I'm most proud of

I guess there are loads, for different reasons, but sonically, maybe Matt Corby's *Telluric*. When I think of that my first thought is always, "That was a great record to work on." It was fun, everyone was so nice and happy. I feel like I learned a lot as a mixer, because it was quite near the beginning of my career as an independent mixer. I was a bit nervous, thinking, "Am I doing the right thing? Am I making the right choices?" I really wanted to make it good, to make an impact. So I wasn't playing it safe. I was like, "No, I want them to like, be blown away by this." I just remember really spending the time getting everything right in that record. I think it paid off because people said that record sounded amazing. I'm not saying I don't do that now, but when you get busy it can become more like a conveyor belt, where you just want to make it sound good and make them happy. I find it harder to get a project that I can really get into and spend a lot of time on every every element to make it sound amazing. It does happen, but having a decent budget and that creative freedom is a luxury.

So *Telluric* was a very good record to work on, personally. And it is also one that people keep coming back to me about, asking, "How did you do the drums on this one? How did you do the guitar?" People seem to like it. Sometimes you get stuff that is already, like, 80 percent there, and you've just got to massage it a little bit. This was one where they were like, "Have fun with it! Do your thing!" They gave me a lot of creative freedom, Matt and the producer [*Dann Hume*]. It was super fun to work on, and I'm proud of the way it sounds.

The first thing I look for in a studio

If I'm gonna be recording, then — more than the gear — the instruments. And the space itself, how it's set up. Because I think that's going to make the session flow better. You know, if they have eight 1176s and super-posh speakers, that will certainly help, and that will make it fun for me, playing around with compressors when we're recording drums; but if they have a Rhodes or Wurlitzer, some synths, a bunch of snares that we can try out... I think that's gonna have a more important impact on the sound of the record that we're working on. More than a bunch of compressors and reverbs.

And they need to already be set up for easy access, where it's not like; "Yeah, we have a piano but we'll have to move it and set it up." Because doubt creeps in really quickly. If it takes time to set up, it's easy to say, "Oh, actually, maybe we don't need that extra guitar." You want to be able to just put a couple of mics on and get on it, because that could completely change the direction of the song and it could be something beautiful. It could be something horrible as well, but at least we tried it! I find sometimes that musicians work like that, where they have an idea and say, "Why don't we try something in this verse? That synth or something like that?" And if you already have like four synths set up there, that they

Photo: Tommy Pullen

can jump on, I think that keeps the session flowing very organically. No one gets bored and starts just scrolling on Instagram.

The person I would consider my mentor
Craig Silvey. I'd worked with a lot of other producers and mixers before him, but with Craig it was such a relationship — it was four years. We used to joke that we saw each other more than our significant others! But it was so cool. I have so much respect for him. Craig's very open to collaboration and to suggestions. I was learning from him, but I was trying to bring my own contribution, and he was open to that. He was so open to sharing knowledge, all the time. If he was working on a record, and I'd ask him, like, "Dude, what did you do to that snare?" or something, he'd say, "Just have a look at the session!" He was not shy or protective over any 'secrets'. He was always open to sharing. And he was always very encouraging. At weekends, if he wasn't in the studio and he hadn't left a mix in progress, he'd say I was more than welcome to try my own stuff. And that's how I learned most of the stuff that I guess makes me, me. Using the downtime to try my own mixes, to try different effects on the vocals and see which one works for me. To make mistakes. So I really thank him for that. And he's still a friend.

My go-to reference track or album
If I'm mixing, I wouldn't call it a reference track, because a reference track would change depending on what I'm working on. But a track that still sounds amazing — on every system — is 'Witchita Lineman' by Glen Campbell. I mean, it's an amazing song, but I'm not going to reference the songwriting because I'm not a songwriter! But sonics-wise, it has so many layers, and everything's crystal clear. The strings sound so crispy. His vocal sounds amazing. The drums, the bass line: to me, everything in that song is perfectly balanced. And it was done ages ago! It's not like it's a modern record! If I'm in a new studio, or starting a new project or if I'm going to a new room and just I need to hear something, I know that song inside out.

My top tip for a successful session
Comfort. As in, making people feel comfortable and that they're in a safe environment. If they're not having a good time, it's going to be harder to convince them to try new ideas. If people become uninterested and start scrolling on their phone, they become less likely to participate and bring ideas to the table. It's not a good idea to start the session with full-on hard work. It's more about just having casual conversation and making sure everyone's comfortable. And then we can start with the hard work; you know, go in and do six takes of solid drums. Sometimes band dynamics can be tricky; maybe there's a band member who is more outspoken than other members, but make everyone comfortable and I think that makes the session super easy and makes everything better. It makes my job easier and their job easier.

The studio session I wish I'd witnessed
Michael Jackson, the *Bad* sessions. It would have been great to be there, just a fly on the wall. From the writing session to the mastering, it went through so many stages. The recording would have been amazing, but also the mixing. There are so many stories, so many rumours about those sessions. And I want to believe that all of them are true, because they sound super cool!

The producer I'd most like to work with
Shawn Everett. He's a really quirky character, which I love. One of my most loved records from the last, 10-15 years is *Sound And Colour* by Alabama Shakes. It was like, "Jesus Christ, how did they do this?" So yeah, I'd love to work with him on a record, just to know how he does things. I've heard that he has a very unconventional way of mixing. Like, he puts a low-pass filter at something like 250Hz and just mixes the low end of the whole mix, and then he opens up the track then mixes the top end. I don't know if I could do that. That sounds insane. But I would love to work with him just to see how he does it.

The part of music creation I enjoy the most
Mixing! No question. I love recording, I love producing. But mixing... you can just go in with fresh ears and be like, "Right, I can hear what they the artist and the producer were trying to do. Now let me take it to that next level." So to me, mixing is the most rewarding stage.

I used to think that I wanted to do mixing for a bit and then slowly transition to production. But the more I do mixing, the more I just want to be the best mixer I can be, and be known as a mixer. I know that people are better producers than me. But I'm really comfortable with mixing, and I know what I can bring to the table. It's super fun to do it as well!

The advice I'd give myself of 10 years ago
Trust your gut. And make as many friends as you can, make as many connections as you can. Speak to people. Don't be shy to speak up and show your worth. At the beginning of my career, sometimes I was a bit more reserved than I should have been. Not trusting what I could do. Obviously, with time that goes away, but yeah: trust your gut. If you have an idea, go for it without fear.

Eduardo De La Paz: "Doubt creeps in really quickly. If it takes time to set up, it's easy to say, 'Oh, actually, maybe we don't need that extra guitar.'"

How I Got THAT SOUND

David Kosten ▪ Bat For Lashes ‘Seal Jubilee’

JOE MATERA

English producer and engineer David Kosten has worked with a broad spectrum of musical artists, from Snow Patrol and Keane to Bat For Lashes and Brooke Fraser. He is also a respected musician in his own right, recording and releasing contemporary electronica under the moniker of Faultline. Over the years, he’s helped create many weird and wonderful sounds, and selects ‘Seal Jubilee’ by Bat For Lashes as a favourite.

Gold Standards

“This is my favourite sound and not because it’s a hugely unique or special tone, or involved a complex chain of effects to create, but mainly because of its significance and meaning to me. It signalled a moment in time when I finally, after many years of trying without success, found myself working as a producer on music that felt really brave and creative, and which also found a much wider audience.

“The sound is the crumpled tape, lo-fi main guitar part in this song. We worked very fast on the album this track is home to, 2006’s *Fur And Gold*, and whereas four of the tracks were from our initial demo session recorded at a song per day, ‘Seal Jubilee’ had enough time lavished on it to let us really think about the mood and instrumentation we wanted.

“We’d recorded the guitar initially on a fantastic valve microphone. It now sounded like someone was captured performing right in front of you, a perfect representation of the real thing. But what we really wanted for this was something much more dreamlike and fragile, and being a ‘good’ recording was of no importance.”

For Better Or Worse

“The sound on the track was made by running the guitar into my unserviced, unreliable and often useless 1977 Roland RE-301 Chorus Echo, with a tape that had been on the machine for years. Initially we’d thought about putting an old-style slapback echo on the guitar, but that didn’t feel right, and then someone accidentally soloed the wet-only echo element, and suddenly there was the guitar sound we wanted. You could hear the join of the tape loop crossing the playback head, the tape that had been chewed up multiple times and then salvaged again wobbling and sticking, sounding like a lost recording from decades earlier. I just needed to time-adjust it so it started at the same place as the original guitar and to mute the original guitar completely.

“By doing this, it completely changed the tone of the song, and pointed a way forward for both the song and the rest of the album and probably a career’s worth of me upsetting artists with new ways to make recordings sound much worse. Recording an orchestra at Abbey Road? Can we use the Dictaphone mic please? That pristine vocal sound? Let’s put it through a guitar amp! Surely the drums will be better if we put them through the Moog? Now of course this capability kind of comes in a $20 plug-in — but there’s fun to be had playing with real machines, when you can, that’s very hard to match.”

Hear The Sound

W www.youtube.com/watch?v=XIep1vFl1b8

W https://open.spotify.com/track/3wiQZuzyyuZBXFpgvjdw0Q

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★★★★★

Composer Max Richter's works encompass solo albums, concert hall and theatre performances, ballet, film and TV, including scores for *Hostiles*, *Black Mirror*, *Taboo*, *Mary Queen Of Scots* and *Ad Astra*. This prolific creator now turns his hand to sample libraries with Max Richter Piano, the first of a series of instrument collections released on the composer's SRM Sounds label. The library is 98GB installed and requires Kontakt Player or Kontakt version 6.8.0 or higher.

Recorded at Studio Richter Mahr, Oxfordshire in collaboration with UK company Song Athletics, the product has impressive specs: the piano is a world-beating Steinway D-274 concert grand, while the acoustically isolated studio's near-zero noise floor is ideal for savouring the lingering die-away of note tails. Such subtlety invites a light touch — as Richter points out, "This piano is a specialist. It loves to play quietly and whisper in your ear. If you are looking to cut through a busy mix, then go elsewhere, but if you want to evoke the intimate stories we all carry inside us, then you will be well served by this instrument."

The library contains two deeply multi-sampled versions of the piano. The first is a conventional open miking; the second a darker, more intimate presentation achieved by placing mics underneath the piano, closing its lid and wrapping it in its large padded cover. Both versions feature 10 dynamic layers covering the ppp to mf range, five round robins and three mic positions: the open piano was close-miked with vintage and modern ribbon and capacitor microphones, while the closed version provides close, room and ambient positions.

A quick improvisation confirms this is a very fine instrument. The bass notes' wonderfully deep, richly resonant and imperious tone provide a perfect foundation for the crystalline high register, and although I missed the incision of a loud forte layer in the upper octaves, the piano is a pleasure to play nonetheless. I found it helpful to reduce the default velocity sensitivity control setting to suit my touch, and though I usually favour an open piano sound, I was pleasantly surprised by the close piano's warm, inviting tone and musical adaptability.

The elegantly simple interface sports action, damper and pedal volume controls governing the extraneous mechanical noises associated with piano performance — I prefer not to hear them, but a hint of pedal clunk arguably adds realism. Long and short reverb controls introduce a pleasant artificial ambience, and there's a round robin reset button which currently lacks MIDI automation.

Subtle rather than banging, softly spoken and lacking the digital effects which adorn today's Kontakt libraries, this Steinway D reflects its owner's musical personality. Using his beloved instrument's built-in Spirio player piano system, Mr Richter can record and play back his performances — now, in exchange for 150 quid, so can you.

Dave Stewart

$155

www.srmsounds.com

Excite Audio
Bloom Vocal Aether
Plug-in Instrument

★★★★

Supporting all the main plug-in formats, Bloom Aether Vocal is based on samples of short vocal phrases, both male and female, that can be manipulated in creative ways. The instrument comes with 250 presets based on 112 samples. On opening the plug-in, the window shows a preset selector and four macro controls: Warmer (saturation), Speaker (radio-type sounds), Doubler (stereo ADT) and the Etherial delay/reverb effect. A 3D wave graphic appears in the centre. At the bottom of the window you can see where the samples are loaded, these covering two octaves of the controller keyboard.

The white keys play the samples, and the first five black keys bring in effects while the second five introduce sequenced phrases. The effects keys control playback speed, which can be halved or doubled without changing pitch; they can reverse playback or transpose up or down by an octave. The sequenced effect takes existing samples and manipulates them in volume and/or pitch to produce something new. To ensure the samples fit into any composition, the key can be transposed to any major or minor key.

A dice button allows random samples to be loaded while keeping the sequencing settings the same. Clicking the knob to the right of the dice opens knobs for Delay, Mod, Reverb, Lo-cut and Hi-cut. Clicking the icon to the right of the transpose section opens up the sample banks with drag-and-drop loading.

In the Edit view you see the waveforms, the phrases sequencer (up to 16 steps), envelope and pitch/formant controls as well as controls for the individual effects, the master output and playback speed. The start and stop location of each sample can be adjusted with snap to transients, grid or off. Looping and reverse can also be set here along with a choice of trigger modes.

Now the important bit — the sounds. The vocal phrases are mainly vowel sounds spanning just a handful of notes, making it easy to fit them into your own composition. They can sound quite natural or come over as processed and otherworldly depending on how they are pitched and whether or not the playback speed is changed, The sequences can also have an endearingly unreal quality to them. In some respects, the way the sounds are controlled draws some parallels with Output's Arcade, though these are very different products in other respects. Add a generous helping of reverb and doubling to a voice and it takes on a very ethereal quality, and if you want that far-away transistor radio sound, the Speaker knob will do that for you. Overall a flexible and fun way to generate atmospheric vocal sounds, whether simply picking presets or diving in and creating your own patches. *Paul White*

$59

www.excite-audio.com

Zero-G

Space

Kontakt Instrument

★★★★

Sample libraries or virtual instruments with the word 'space' in the title could probably go down one of two routes; sounds for large open spaces here on earth or sounds for large open spaces in... well, actual space. Given the graphics behind the UI of Zero-G's Space, there are no prizes for guessing which route you get here. In short, if you require a collection of textural sound elements that might enhance your latest sci-fi based film, video game or TV drama, you're in the right place.

Designed for the full version of Kontakt (6.7.1 or later, although you also get all the samples in both WAV and AIFF formats), once installed, Space delivers some 4.5GB of sample content. For Kontakt, these have been organised across 27 individual NKI instruments, each of which is themed to provide a particular type of sound. Via a simple browser within the UI, each of these instruments then gives you access to multiple presets (720 in total across the whole library) within the target theme. The instrument names give a pretty clear idea of what to expect — for example, Astral Pads, Cosmic Rhythms, Deeps Bass (ho ho!), Galactic Synths, Scary Monsters, Drones, UFOs & Aliens, Steller Hits, Wonders Of The Cosmos — and, when you select an individual preset within an instrument, this loads one of the underlying samples (many of these are impressively long) and an effects configuration.

Alien graphics aside, the UI is very straightforward, with a main page providing access to Amp and Filter sections, each with their own envelope and LFO options. Beneath the preset selector/waveform display, there are also buttons to switch each of the effect modules on/off. A second screen (accessed via the FX Rack tab at the bottom of the UI) switches to the full effects control set with plenty of editing and sound-design potential.

While there is perhaps something just a little old-school about the design of the instrument and the nature of the sample playback (there doesn't seem to be any time-stretching applied as the sample is triggered at different pitches, for example), the sounds themselves are rather impressive. Yes, the library is as much about adding sound-design elements, textures or atmospheric drones to your overall soundtrack as it is creating the more melodic or harmonic content of a score but, with that duly noted, there are plenty of elements here that would happily complement some epic external spaceship footage, aliens in the shadows, or something big hitting something else big in a vacuum (I know that wouldn't actually make a sound, but that doesn't stop sound designers adding it).

There are, of course, other libraries and/or sound-design tools that can generate these types of sounds. However, for simplicity of use, and for a very modest price, in sonic terms, Zero-G's Space punches well above its weight.

John Walden

$54.95

www.zero-g.co.uk

Sonora Cinematic

Aria Vocalscapes

Kontakt Instrument

★★★★★

Sonora Cinematic have an enviable track record when it comes to the creation of interesting Kontakt instruments. Continuing their philosophy that begins with great samples, Aria Vocalscapes contains some beautiful content performed by classical soprano Emma May Price. However, this does not set a stylistic precedent for the instrument.

Patch creation begins with the option to load up to two sample partials, described as A and B. These are clearly represented, with the central presence of an X/Y pad that can be used to manually create movement between the two partials. This movement can be recorded and replayed, and sync'ed to your DAW.

The samples themselves are presented in a menu format, with a considerable amount of variety within the vocal remit. The 39 available samples range from predictable 'oohs' and 'aahs' to the African-inspired Tikiti. There are whispers and 'shh' samples, as well as examples that have been treated electronically with effects such as granular synthesis or cassette-style degradation. In all scenarios the sampling is up-close and personal in a one-shot capacity, although this latter point is not really an issue, thanks to the considerable processing power of the plug-in.

The resynthesis and interaction between the two partials is at the heart of the instrument and really helps to create depth and movement. Apart from the aforementioned X/Y pad, a dedicated page reveals LFO and filter layers, which can be applied to each individual sample. The LFO can be directed to the filter cutoff, volume and frequency, and has five waveforms to choose from. The filter provides cutoff and resonance control, and can also be linked to an independent step sequencer, providing up to 16 steps of automated control.

This instrument undoubtedly excels in the creation of soundscapes, pads and audio backdrops, and this is in no small part due to the extensive effects section. The effects call upon amp simulations, distortion, saturation, delay, reverb and a large number of modulation effects, including ring modulation. With three flexible slots, you can dictate the effects chain to suit your own design. In line with its pad-like and cinematic tendencies, the delay and reverb elements can be cranked up to a pretty merciless state, although in the case of the delay the presence of a continued repeat can be somewhat blurred, allowing simple elongation of your chosen vocal sound.

Aria Vocalscapes is an inspiring and creative product to use, thanks to its diversity and musicality, which stems from the initial sample content. It ships with a large number of presets, but for my money, it's at its most creative when you dive in to the instrument and edit or create your own palette.

Dave Gale

$74

www.sonoracinematic.com

[Audio examples of this month's libraries are available at www.soundonsound.com.]

Q&A

QUESTIONS ANSWERED, PROBLEMS SOLVED

Q Can I calculate the preamp gain difference between two mics?

I have two different mics, made by the same legacy German manufacturer, that I'm 'shooting out' to compare them as vocal mics, and generally to make sure they're working properly. They're amplified by two separate but identical preamps and I'm noticing that, to get the same level coming into the DAW from each, one preamp needs almost exactly 10dB more gain. Is there a way to calculate, from the published sensitivity specs, whether this discrepancy in gain is correct — or is it not so simple? Here are the published specs for the two mics:

	Mic 1	Mic 2
Max SPL	127dB	125dB
S/N Ratio	82dB	74dB
Sensitivity	28mV/Pa	8mV/Pa

SOS Forum post

Hugh Robjohns, Technical Editor

The only relevant specs for matching output levels are the sensitivity numbers: the higher the sensitivity, the larger the output level for a given source. When determining sensitivity, the acoustic reference is normally 1 Pascal (Pa) of air pressure, which equates to a sound pressure level of 94dB SPL. With that sound pressure acting on the mic's diaphragm, the generated output voltage is measured in milliVolts, hence the sensitivity being given as mV/Pa. Your mics are specified as 28mV/Pa and 8mV/Pa and to calculate the difference in output level, you use the standard formula: dB = 20 x Log (v1/v2). The figures from your mics give us: 28/8 = 3.5. And since Log 3.5 = 0.544, we get: 20 x 0.544 = 10.881dB. Assuming the published specs are accurate, then, to perfectly match the output level of the two mics the quieter one must be amplified by 10.9dB more than the louder one. Your estimate of 10dB is very close!

Given the sensitivity of two mics, you can calculate precisely the preamp gain difference required to raise their signals to the same level.

Q Why does my kick lose weight when I compress the drum bus?

I've noticed that quite a few engineers who work with bands like driving their drum submix into fairly heavy compression, but whenever I try doing this it seems like I end up losing the low-end weight from my kick drum. Why is this happening — and is there anything I can do to avoid it?

Anon via email

Mike Senior, mixing expert

There are a couple of reasons why a bus compressor can rob your drum mix of low-end weight, and I'll deal with them one at a time.

Probably the most common cause is that, on the whole, compressors are more sensitive to low frequencies than high frequencies. This is why, for example, vocal compression often overemphasises sibilant consonants — the lower-frequency sung notes trigger more gain-reduction than the higher-frequency esses do. With mixed drum signals, this frequency bias means a strong kick will usually trigger more gain reduction than the other kit instruments, because it has more low-spectrum energy. If you've set up your compressor to release its gain reduction reasonably fast, then the kick will effectively be turned down more than snare and tom hits, giving the impression that your drum kit as a whole is losing low-end weight.

It might seem pretty counterintuitive, but one thing that can remedy this problem is high-pass filtering the kick-drum channel. Yes, you read that right: cutting the kick's low end! Specifically, you should cut only the very lowest frequencies, and keep the filter's cutoff frequency low enough that you scarcely hear any difference to the soloed kick-drum channel. The reason this can improve matters is that many project-studio drum submixes have flabby subsonic frequencies lurking underneath the kick drum. These frequencies won't be contributing anything subjectively useful to the kick-drum sound at all, and may even be too low for you to hear if you're listening on headphones or typical small project-studio monitors. Crucially, though, these subs may still merrily trigger excessive gain reduction from your compressor. By filtering them out, your kick appears to the compressor to have less low-frequency energy, so it triggers less gain reduction per kick hit, leaving your drum submix sounding weightier.

The simplest way to prevent drum mix compression robbing your kick of low end is to use your compressor's side-chain high-pass filter.

You have to be careful high-pass filtering any drum channel, though, as the time-smearing and resonant effects of the EQ can potentially soften your attack transients. So do keep your ears peeled for that side-effect. If you encounter this problem, there's another method of reducing the compressor's kick-drum gain-reduction if your compressor includes side-chain EQ functionality: just cut some low end from the compressor's detector signal rather than the drums themselves. This effectively makes the compressor less sensitive to low frequencies, so it won't respond as strongly to the kick-drum hits.

The second reason a compressor can cause a drum mix to lose low end is that you're using a very fast-acting compression setting — the rapid gain reduction change triggered by each kick hit can put an appreciable kink in the hit's waveform, rather than just shaping its level envelope. In this situation, the slower-moving low-frequency waveform cycles will inevitably be more heavily waveform-mangled than the faster-moving high-frequency ones, so the result is that the low frequencies get distorted (effectively converting some of their energy into upper-spectrum harmonics). Probably the best protection against this is simply being aware that it can happen. After all, it's pretty simple to increase your compressor's attack time a little once you've realised it's the source of the problem. Of course, lengthening the attack time may lessen the compressor's gain reduction action across the board a little, and you may have to adjust your threshold and ratio settings to compensate if that's a concern.

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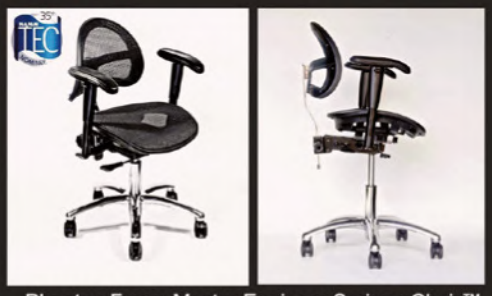
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SOS WHY I LOVE...

VINYL

NELL HOUGHTON

Last year I watched *Almost Famous*, and that film made me realise something incredibly sad: I'd never heard anything on vinyl. You see, I'm 12. My friends and I have grown up in a world of smartphones, headphones and playlists, where we can listen to anything we want, anywhere and at any time, and where technology tells us what to play next.

But music means so much to me that I wanted to try listening in a different way, and I decided I'd explore the world of vinyl. I begged my parents for a hi-fi separates system, and last Christmas, the fat guy with a beard (sorry Dad!) actually delivered it. I wouldn't trade it in for the world! It's made up of second-hand bits (I prefer 'vintage'), but with a Technics CD player, an Akai turntable, a NAD amplifier (thanks Hugh Robjohns!) and a lovely pair of Mordaunt-Short speakers, I think it's an extraordinary first hi-fi.

As with any kid of my generation, I'm into lots of modern music and I haven't abandoned Spotify and YouTube. But a big reason I wanted a record player was to gain a sense of what it felt like listening to music at a time when it seems to have been that much more special to people. So, I started with records by older groups: REM, the Verve, the Lemonheads and, my pride and joy, *Led Zep IV*. Recently, I went to my first vinyl fair and sniffed out some more bargains, including an imported green vinyl version of *Led Zep II*. That one not only sounds great, but it's beautiful too!

I realise vinyl isn't all good. It's easy to scratch or break it, it's incredibly complicated to use, the sound isn't always perfect, and records are expensive. But it's all about the experience and the connection with the music, and the first time I listened to a record I was completely blown away. It wasn't only me, either: I played a couple of records when my bestie came over, and the expression on her face said everything. I'd never seen anyone look so insanely gobsmacked: speechless and fascinated, hypnotised for a good two minutes, she eventually managed to say "But... how... does it... work?" She's not alone. Most of my generation is missing out on this joy of just sitting down and absorbing good music.

Most parents might not think to spend hundreds of pounds on a hi-fi when their children could just use their phone or iPad. But it's worth it! With no screen to distract you, you have to just sit back and close your eyes, or explore the cover art and lyrics. You get swept up in the music, and appreciate it in a way you just can't if listening on your phone while you do your maths homework! So, remember, if you win the lottery, don't give it all to charity: put some aside to buy a kid a record player...

Naturally, I've had Spotify playing the whole time I've been writing this. But I'm making an exception, as I'm listening to Billie Eilish, and nothing says 'good music' to me better than that. And Billie, if by any chance you're reading this and feeling kind, could you possibly spare two tickets to one of your concerts for an adoring fan?

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